(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 0 480 010 B1

(12) EUROPEAN PATENT SPECIFICATION

(45) Date of publication and mention of the grant of the patent:
11.09.1996 Bulletin 1996/37

(21) Application number: **91908882.3**

(22) Date of filing: **02.05.1991**

(51) Int Cl.⁶: **G10L 5/06**, G10L 5/00

(86) International application number:
PCT/US91/02907

(87) International publication number:
WO 91/17540 (14.11.1991 Gazette 1991/26)

(54) **SIGNAL RECOGNITION SYSTEM AND METHOD**

VERFAHREN UND VORRICHTUNG ZUR SIGNALERKENNUNG

SYSTEME ET PROCEDE DE RECONNAISSANCE DE SIGNAUX

(84) Designated Contracting States:
AT BE CH DE DK ES FR GB GR IT LI LU NL SE

(30) Priority: **02.05.1990 US 517835**

(43) Date of publication of application:
15.04.1992 Bulletin 1992/16

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(56) References cited:

EP-A- 0 283 743	US-A- 4 677 466
US-A- 4 697 209	US-A- 4 843 562

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Description

BACKGROUND OF THE INVENTION

This invention relates to apparatus and method for recognizing signals, and in particular to apparatus and method for recognizing signals by utilizing statistical moments of sampled signal values to produce feature vectors, and to quantization of the feature vectors in order to compare the signal to a predetermined signal data base, and to derive the signal data base.

While the present invention will be described with respect to a system for recognizing broadcast signals such as music, it is to be understood that the teachings of this application are applicable to a broad spectrum of signal recognition fields.

The accurate recognition of broadcast signals is important to marketing executives, royalty collection agencies, music promoters, etc. It is well known that a wide variety of legal, economic, and social concerns require the regular monitoring of broadcast information. All such requirements share a common need for certain information such as which information is broadcast and when. In the prior art, broadcast stations were monitored manually by a plurality of listeners who would physically monitor the broadcast program and manually tabulate which information was broadcast at what time. Problems of reliability and cost have stimulated the effort toward realizing automated broadcast signal recognition systems. An initial automated method included encoding a unique cue signal in each song, and then monitoring each broadcast station to detect the cue signal. However, the associated encoding and decoding circuitry is expensive and complicated, and government regulatory agencies are adverse to providing additional bandwidth necessary for providing a large plurality of unique cue signals.

A further advance in the field of automated broadcast signal recognition is disclosed in U.S. Patent No. 3,919,479 to Moon et al. In Moon et al., an audio signal is digitally sampled to provide a reference signal segment which is stored in a reference library. Then, when the audio signal is broadcast, successive portions thereof are digitized and compared with the reference segment in the library. The comparison is carried out in a correlation process which produces a correlation function signal. If the reference and broadcast signal segments are not the same, a correlation function with a relatively small amplitude results. On the other hand, if the reference and broadcast signal segments are relatively the same, a large correlation function signal is produced. The amplitude of the correlation function signal is sensed to provide a recognition signal when the amplitude exceeds a predetermined threshold level.

However, the single segment correlation system of Moon et al. is subject to signal drop-out which may disable the system altogether. Also, the Moon et al. system is relatively susceptible to time-axis variations in the broadcast information itself. For example, it is known that many disc-jockeys "compress" broadcast songs by speeding-up the drive mechanism. It is also known that other disc-jockeys regularly "compress" and/or stretch" broadcast information to produce certain desired effects in the audience. Moon et al. attempts to overcome such time-axis variations by reducing the bandwidth of the broadcast signal by envelope-detecting the broadcast signal and providing envelope signals having substantially low, and preferably sub-audio, frequency signal components. It has been discovered that when the envelope signal at sub-audio frequencies is used during the correlation process, the digitally sampled waveforms are less sensitive to time-axis variations. However, the improvements which can be achieved by such a solution are very limited and will only operate for broadcast signals which have been "compressed" or "stretched" by a small amount. In addition, such a solution is subject to high false alarm rates. These disadvantages make the Moon et al. system less than desirable for a rapid, accurate, and inexpensive automatic broadcast signal recognition system.

A further advance in the automatic signal recognition field is disclosed in U.S. Patent No. 4,450,531 to Kenyon et al. The some Mr. Kenyon is the sole inventor of the subject application. The system of the '531 patent successfully addresses the reliability problems of a single segment correlation system and the time-axis variation problems experienced by prior systems. In the '531 patent, a plurality of reference signal segments are extracted from a program unit (song), digitized, Fourier transformed, and stored in a reference library in a frequency domain complex spectrum. The received broadcast signal is then prefiltered to select a frequency portion of the audio spectrum that has stable characteristics for discrimination. After further filtering and conversion to a digital signal, the broadcast signal is Fourier transformed and subjected to a complex multiplication process with reference signal segments to obtain a vector product. The results of the complex multiplication process are then subjected to an inverse Fourier transformation step to obtain a correlation function which has been transformed from the frequency to the time domain. This correlation function is then normalized and the correlation peak for each segment is selected and the peak spacing is compared with segment length. Simultaneously, the RMS power of the segment coincident with the correlation peak segment is sensed to determine the segment power point pattern. Thus, the '531 patent overcomes the disadvantages of a single segment correlation system by providing a plurality of correlation segments and measuring the distances between the correlation peaks. Where the distances match, the broadcast signal is declared as being similar to the signal segment stored in the reference library. In addition, the RMS value comparison operates to confirm the classification carried out using the signal segments.

To overcome the time-axis variation problem, the '531 patent utilizes an envelope detector and a bandpass filter for the broadcast information. However, the system according to the '531 patent is computationally very demanding. For example, performing the various multi-segment correlations requires a great deal of computer power. Since a multitude of segments are sampled, the system according to the '531 patent may take a good deal of time and require the use of expensive, powerful computers.

An automated approach to speech pattern recognition is disclosed in U.S. Patent No. 4,282,403 to Sakoe. Sakoe discloses a speech recognition system in which a time sequence input of pattern feature vectors is inputted into a reference library. The received speech signal is then subjected to spectrum analysis, sampling, and digitization in order to be transformed into a time sequence of vectors representative of features of the speech sound at respective sampling instances. A time warping function may be used for each reference pattern by the use of feature vector components of a few channels. The time warping function for each reference pattern feature vector is used to correlate the input pattern feature vector and the reference pattern feature vector. The input pattern feature vector sequence is then compared with the reference pattern feature vector sequence, with reference to the time warping function, in order to identify the spoken word. However, the Sakoe system time warps the reference patterns rather than the input signal, and thus a plurality of patterns must be calculated for each reference pattern thus increasing the memory and computational demands of the system.

A further signal recognition system is disclosed in U.S. Patent No. 4,432,096 to Bunge. In Bunge, sounds or speech signals are converted into an electrical signal and broken down into several spectrum components in a filter bank. These components are then integrated over a short period of time to produce the short-time spectrum of the signal. The spectral components of the signal are applied to a number of pattern detectors which apply an output signal only if the short-time spectrum corresponds to the pattern adjusted in the relevant pattern detector. Each pattern detector has two threshold detectors which supply a signal if the applied input lies between the adjustable thresholds. Thus, the pattern detectors supply an output signal only if all threshold value detectors are activated. For each sound of speech, a pattern detector is provided. When a series of sounds is recognized, the series of addresses of the pattern detectors which have successfully generated an output signal are stored and subsequently applied to the computer for comparison. It can be readily appreciated that such a system requires a number of pattern detectors and a corresponding powerful computation device. In addition, while the Bunge system uses a filter bank to provide a low frequency output signal which is relatively less sensitive to time-axis variations, the Bunge system is still subject to time distortion problems and a high false alarm rate.

A recently commercialized automatic broadcast signal recognition system is disclosed in U.S. Patent 4,843,562 to Kenyon et al. Again, the same Mr. Kenyon is the sole inventor of the subject application. In fact, specific teachings from the '562 patent will be incorporated in further portions of this specification. The '562 patent describes a two-stage (coarse and fine) classification system using fewer processor resources. According to the '562 patent, the broadcast signal is bandpass filtered, rectified, and lowpass filtered to provide a plurality of low bandwidth waveforms. The waveforms are sampled and the samples are used to generate a spectragram which is then compared with a plurality of reference spectragrams stored in a first stage reference library. The first stage reference spectragrams are then queued in order of their similarity to the generated spectragram. Next, a plurality of second stage reference patterns, which correspond to the queued first stage reference spectragrams, are correlated with one of the analyzed waveforms in the queueing order established previously. A correlation value is provided for each second stage reference pattern stored in the second stage reference library. When it is determined that a correlation value exceeds a threshold value, a recognition is declared and the broadcast signal is classified as similar to the second stage reference pattern whose correlation value exceeds the threshold. The analyzed waveform used in the second stage classification is time warped to account for speed fluctuations in the broadcast signal.

While the system according to the '562 patent is successful, it is somewhat limited in its ability of recognizing a large number of songs. For example, the system according to the '562 patent is capable of recognizing any of 600 songs on a single channel with high reliability. The system can simultaneously monitor 5 different channels. However, a system which could identify any one of three thousand songs on each of five simultaneously broadcast stations with high reliability would provide a very attractive and commercially successful signal recognition system. Further, the system according to the '562 patent required approximately 64 seconds to detect and classify a broadcast song. It is desired to reduce this time to 28 seconds to allow for the identification of shorter duration recordings such as advertisements. While increasing performance, it is important to retain the desirable compact architecture of the '562 patent.

Thus, what is needed is an improved system for accurately recognizing and classifying a large number of unique broadcast signals on a plurality of broadcast channels simultaneously and with high reliability. The system must be small, inexpensive, and easy to operate.

SUMMARY OF THE INVENTION

The present invention is designed to overcome the disadvantages of known automatic broadcast signal recognition

systems while at the same time satisfying the objectives discussed above. Furthermore, the present invention has application to a wide variety of signal recognition fields, and not just to the recognition of broadcast signals.

The present inventor has discovered an improved method of signal recognition in which a signal is received and sampled at a plurality of sampling points to produce a plurality of signal values. A statistical moment of the signal values is then calculated according to the formula:

$$(1/N) \sum_{n} \frac{(X(n)-\mu)^k}{\sigma^k}$$

where:

N = the number of sampling points;

n = 1<n<N;

X = the signal value of said signal at the sampling point;

μ = a mean of the signal values;

σ = a standard deviation of the signal values

k = an integer greater than 1;

The calculated statistical moment is compared with a plurality of stored signal identifications and the received signal is then recognized as being similar to one of the stored signal identifications.

Preferably, the received signal is bandpass filtered, rectified, and lowpass filtered to provide a plurality of low frequency waveforms. Then, the low frequency waveforms are combined into a plurality of linear combinations thereof. Each linear combination is then sampled to produce the plurality of signal values which are used to calculate the statistical moment.

Preferably, two statistical moments (skew and kurtosis) are calculated for the sampled values of each linear combination. A plurality of feature vectors may then be derived, each feature vector comprising the skew and kurtosis values for all linear combinations within a predetermined sampling time period.

Each feature vector may then be quantized by replacing the floating point values of skew and kurtosis with single integers in accordance with a predetermined quantization pattern. After quantization, a weighted sum of the quantized vector may be calculated using a non-decimal radix. The weighted sum value may then be used to address a data base which stores signal identifications in accordance with the address.

The present invention also proposes a system for creating the signal data base which is accessed in order to recognize the received signal. The process of creating the data base is very similar to the process of recognizing the signal noted above. However, the target signal is first analyzed to determine a spectrally distinct portion thereof. The spectrally distinct portion of the signal is then bandpass filtered, rectified, and lowpass filtered to produce the low frequency waveforms from which a plurality of linear combinations are derived. The linear combinations of the spectrally distinct portion are then subjected to a modified sampling procedure whereby each linear combination is sampled a number of times with a moving window. The sampled values are then used to calculate skew and kurtosis, thus producing a plurality of reference feature vectors for the spectrally distinct portion of the signal.

The plurality of feature vectors are quantized in order to make their values more distinctive by spreading the distances between the vectors in the feature hyperspace. Two procedures for quantization are possible. First, a non-overlap quantization scheme can be adopted whereby the feature vector signal value population is evenly divided into a plurality of segments, for example 5. Then, each value of the skew or kurtosis in each feature vector is plotted and assigned one of the five values. Thus, a plurality of quantized vectors are produced for each sampled signal. An overlapping method of quantization is also possible whereby the signal value population of the skew and kurtosis are divided into a plurality of overlapped areas wherein a skew or kurtosis value may be assigned to two areas. Each feature vector is then plotted and two quantized vectors are produced, since each skew or kurtosis value is capable of assuming two values. The two quantized feature vectors are then permutated to provide a further plurality of quantized feature vectors for the reference data base.

Whatever quantizing scheme is adopted, preferably, a weighted sum of each quantized vector is produced using a non-decimal radix. The weighted sum is then used as an address to access a data base wherein a pointer is stored, the pointer pointing to a further data base location where the target signal's identification code is stored.

BRIEF DESCRIPTION OF THE DRAWINGS

The advantageous features of the present invention will be readily understood from the following description of the presently preferred exemplary embodiment when taken together with the attached drawings in which:

Fig. 1 is a block diagram depicting the system according to the presently preferred embodiment;
Fig. 2 is a block diagram showing the principle of filtering the received signal into four different frequency bands;
Fig. 3 depicts a series of waveforms showing the wave-shaping carried out in the processor of Fig. 1;
Fig. 4 is a series of waveforms showing four feature sequences generated by the processor;
Fig. 5 is a chart showing the overlapping sampling of a spectrally distinct portion of the signal for creating the reference libraries.
Fig. 6 depicts the feature vectors generated by the sampling performed in Fig. 5;
Fig. 7 is a vector quantization table showing the number of distinct identities for a given number of quantization levels and a given number of features;
Fig. 8 shows the quantization of a feature vector;
Fig. 9 is a depiction of the quantized feature vector plotted in three-dimensional space;
Fig. 10 depicts the population density of kurtosis values used in non-uniform vector quantization without overlap;
Fig. 11 depicts how the feature vector quantities are assigned quantized values in the population of Fig. 10;
Fig. 12 depicts the population density of kurtosis values used in non-uniform overlap-encoded vector quantization;
Fig. 13 shows how feature vector quantities are assigned quantized values in the population of Fig. 12;
Fig. 14 shows how the feature vector is quantized in the overlap-encoded vector quantization scheme;
Fig. 15 depicts how the quantized vector is converted into a weighted sum which is used to address a data base to identify a signal;
Fig. 16 is a top-level flow chart depicting a method according to the preferred embodiment;
Fig. 17 is a flow chart depicting how the statistical moments are calculated; and
Fig. 18 is a flow chart showing the confirming step according to Fig. 16.

DETAILED DESCRIPTION OF THE PRESENTLY PREFERRED EXEMPLARY EMBODIMENT

While the present invention will be described with the respect to an improvement in the system described in the '562 patent, persons having ordinary skill in this field will recognize that the teachings of this invention may be utilized in a wide variety of signal recognition environments. For example, the present invention will find application in voice processing systems, speech recognition systems, acoustical communications systems, etc.

First, an overview of the invention will be provided for clarification purposes. The '562 patent describes apparatus and method wherein broadcast information is recognized using a two-step classification process. In the first step, the input signal is compared to a first stage library and a coarse spectral analysis is performed. The first stage comparison generates a queue of signal identifications from the first stage reference library according to the coarse spectral analysis. Then, a second, finer correlation is performed in the second stage in the queueing order established in the first stage. The present invention proposes an improvement in the first stage classification process in order to reduce the recognition time and the number of entries in the first-stage queue. In fact, the present invention has demonstrated such high efficiency that it may be useful without the second stage for certain signal recognition systems.

The first stage analysis screens the input signal and eliminates a large number of candidate signals from consideration in the computationally demanding time-warped correlation carried out in the second stage. If the first stage efficiency can be increased from 80% to 90%, the system capacity is doubled. A prototype of the present invention has demonstrated over 98% efficiency in rejecting false alarms. Thus, a factor of 10 improvement in the system according to the '562 patent can be achieved.

In the '531 patent, an efficient signal recognition system was disclosed that maintained a sufficient time-bandwidth product to allow accurate discrimination of signal signatures while providing a means for making the system insensitive to broadcast speed variations and transmission aberrations. This technique, known as segmented correlation, and the other implementation details described in the '531 patent provided a highly successful system capable of recognizing any of 25 songs on a single channel with high reliability. The present invention has as a goal the ability to identify any of six thousand recorded signals on each of five simultaneous broadcast channels with similar reliability. Additional system tuning and prototype modeling should increase the capacity to nearly ten thousand songs per channel.

Central to the concepts embodied in the '562 invention is a technique for compensating for broadcast speed variations known as linear time warped correlation. This technique is more robust than the previous segmented correlation approach in that it maintains a stronger relationship between portions of the feature vector that were derived from different portions of the recording. In addition, a hierarchical recognition procedure serves to multiply the available processing capacity by screening the candidate recognitions using a coarse sorting algorithm. This technique con-

sumes less computing resources to scan the entire signal data base, excluding candidates that clearly will not pass the second-stage recognition criteria. Those that are acceptable by the first stage process (typically 10 to 20%) are analyzed by the computationally intensive linear time warped correlation procedure of the second stage in the order of probability of recognition (queue) established in the first stage. Besides greatly increasing the system capacity, the '562 invention reduced the reference pattern duration from the previous 64 seconds to 28 seconds to allow for the identification of shorter duration signals, such as commercials.

The technical approach leading to the presently proposed system will now be described. Given the system objectives noted above, two problems presented themselves. First, more distinct feature vectors must be provided for the first stage analysis, and second, the new architecture must be compatible with the architecture described in the '562 patent. That is, the first stage classification process must accept the same set of four envelope detected frequency bands used in the '562 patent. The output of the first stage classifier must be a list of pattern numbers to be evaluated in the second-stage correlator.

To be effective, the first-stage algorithms are required to be tolerant of speed variations and other aberrations while providing high discrimination between patterns. The false alarm rate must be minimized without causing missed detections. In fact, the first stage classifier must be biased so that only signatures that are clearly incorrect will be rejected. This usually results in an increase in the false alarm rate. It is difficult to simultaneously optimize both of these performance parameters.

A feature vector may be used to define a feature space having a number of dimensions equal to the number of features in the vector. Such a feature space may be denoted "hyperspace". The feature vectors of each target signal are stored in hyperspace according to their calculated values. In recognition, the feature vector derived from the received signal is used to search hyperspace to discover the signal or signals whose signatures are stored in a particular area of hyperspace. Hyperspace may contain a plurality of "clusters" in which the feature vectors of a plurality of signals may be stored. For example, similar signals may produce similar feature vectors which would be stored in close proximity to each other in hyperspace. If the spacing between such signal clusters in hyperspace is large relative to the size of the clusters, it may be possible to provide a region of uncertainty around each feature vector in hyperspace that has a minimum overlap of other regions therein. One method of accomplishing this increased spacing is to increase the dimensionality of the feature hyperspace. The value of each feature is treated as a coordinate in hyperspace. For example, as depicted in Fig. 9, if three features are used, they form a feature vector that specifies a position in three dimensional space. If the feature vector is then quantized (assigned to a particular category based on the detected feature value), each component can take on one of a number of discrete values, e.g. 5. Then, the space will contain 125 discrete locations. If a signal identification code is assigned to each location, then it is possible to extract and quantize the same features from a received signal, the feature vector can be mapped to the same location in the three dimensional space, and the appropriate signal identifier can be retrieved. This would allow unique access for 125 signals if their feature vector values are unique after quantization. Unfortunately, the features that are measured are not unique. Several different signals could generate the same feature vector. This may be handled by assigning a list of possibilities to each location in hyperspace. Fig. 9 shows an example of a three-dimensional hyperspace wherein each of the three features can take on 5 values. A "cluster" 40 is depicted therein centered on the location (S(4-3)=3; S(2-1)=2; S(3-2)=0).

A second problem is that a signal or pattern to be recognized may not generate the appropriate vector due to time compression or other transmission aberrations. Thus, the variability of the features must be studied, and it must be determined how the resultant feature vectors will designate different regions in hyperspace. The search for distinct features is based on the criteria that they are stable under the influence of expected aberrations and that they are distinctive. If one can assume that the individual features that compose the feature vector are independent, then by increasing the number of features, the separation between clusters in hyperspace can be dramatically increased without increasing the cluster size. To allow for variations in the quantized features, and to reduce the requirement for time alignment, multiple feature vectors can be generated at different time displacements, allowing each signal to sweep out a region of the hyperspace that it will occupy. The search for such distinctive time invariant feature vectors will be described below.

One of the most important aspects of any pattern recognition problem is the identification, creation, and selection of features that are stable when a particular signature is present, but provide different indications if the signature is not present. In developing this invention, audio signals from records, tapes, and airplay were digitized at a rate of 2500 samples per second after being low-pass filtered at a cut-off frequency of 1.0 kHz. The frequencies below 1 kilohertz have been shown to be relatively unaffected by spectral modifications employed by broadcasters.

Approximately 65 seconds of audio from these signals was digitized from 72 songs and stored as an experimental data base. Each of these data files was then processed by a software simulation of the front end processor that forms the envelopes of four frequency bands. The output from the simulator was a set of multiplexed floating point files which could be plotted on a graphics terminal or processed by a program to determine characteristics of the data that could be exploited to form good features for the feature vectors. A spectral analysis program computed the power spectrum,

log power spectrum, cepstrum, and auto-correlation functions of each frequency band. The power spectrum and log power spectrum are frequency domain functions, while the cepstrum and auto-correlation functions are time domain functions. Next, a statistics program was used to compute statistical moments of the signal values of the four bands of the received signal. Such statistical moments include the mean, variance, skew, and kurtosis. In addition, the co-variance matrix of the four bands was computed.

In examining the power spectra and the log power spectra, it was discovered that while different songs had spectral peaks at different frequencies, it was not clear how to select a small number of spectral components that would reliably discriminate among the different songs. This applies to both the linear and log power spectra. However, computing the signal bandwidth of each band provides a set of four numbers that should be stable with regard to speed variations. These features were evaluated utilizing eight different time delays.

In examining the auto-correlation functions, it was discovered that the only stable features that could be derived were closely related to the bandwidth measurements discussed earlier. This is to be expected since the power spectrum and the auto-correlation function are a Fourier transform pair. Features derived from the auto-correlation function were therefore considered to be redundant and were discarded in favor of those derived from the power spectrum. The cepstrum was examined and found to have no stable useful features. This is because the cepstrum exploits the presence of harmonically related signal structures to determine the period of the composite waveform. These harmonic structures were not found to be present in the envelopes of the bandpass filtered audio.

In studying the behavior of the features produced by the statistics program, it was discovered that skew and kurtosis values were distinctive between songs, and were relatively stable when computed at time offsets within the song. There are two other characteristics of these features that were found to be attractive. First, both skew and kurtosis are self-normalizing. That is, their value is independent of loudness. Second, their values are virtually unaffected by speed variations in the recording. This is because their calculation does not explicitly involve time or frequency. This was considered to be quite important since a significant amount of effort has been directed towards making the entire system immune to speed variations induced by broadcasters. It was discovered that some songs produced significantly different values of skew and kurtosis at different time offsets. This occurred when the time offsets moved the window of analysis into regions of a song where the entire character of the song was different. This is assumed to be a general problem that will affect any set of features chosen, and therefore must be dealt with in the recognition algorithm instead of the selection of features to be used.

In conclusion, the present inventor discovered that the properties of skew and kurtosis are well suited for categorizing broadcast signals efficiently and reliably. Bandwidth estimations also performed well, but were less immune to the effects of time offsets. Since the recognition algorithm deals directly with this, envelope bandwidth may still prove to be a useful discriminating feature. It should be pointed out that each of the features discussed was computed for all four frequencies bands. Therefore, a total of 12 features were considered to be qualified in terms of their discrimination capabilities, four for each of skew, kurtosis, and envelope bandwidth. If all 12 features are used, a 12-dimensional feature hyperspace would be produced. This probably would be unacceptable from the point of view of the required memory, and the number of features that must be calculated. Therefore, in terms of the present embodiment, it was decided to use only the skew and kurtosis values, thus producing a total of eight features. However, this still may be unacceptable in view of the required memory. Therefore, the four frequency bands were linearly combined into three waveforms, thus providing skew and kurtosis values for each of three waveforms, producing six dimensional feature hyperspace. A confirmation of this choice will be discussed in greater detail with respect to the vector quantization procedures discussed below.

Vector quantization is in principle a simple concept. It involves reducing the resolution of each vector component from its continuous range of values to one of a small number of values, e.g., reducing a plurality of multi-digit values to a plurality of single-digit values (see Fig. 8). For example, the feature value range could simply be partitioned into several bands of equal width. The feature is then encoded or quantized by replacing its multi-digit value with its band number. There are two primary problems with this simple approach. First, values that fall close to the boundary between two bands may be erroneously quantized due to noise, distortion, or computation precision limitations. Second, it is not likely that the feature values will be uniformly distributed over the entire range. This would lead to a heavy concentration in certain bands, reducing the information content of the quantized feature. In the limiting case where all values fall into the same band, the feature would contain no information at all. This is dealt with by using non-uniform width of the bands (e.g., Fig. 10), and by employing an overlap quantization process for the encoding of reference feature vectors for the reference library (e.g., Fig. 12).

First, it is necessary to consider the number of quantization levels and the number of features employed in the feature vector. These specify the dimensionality and the density of the feature hyperspace. Fig. 7 lists the number of discrete locations in the hyperspace as a function of the number of features used and the number of levels to which they are quantized. Notice that for all cases where the number of levels is greater than 3, more is achieved by increasing the number of features than by increasing the number of quantization levels. It is worthwhile to employ the largest possible feature hyperspace within the limits of memory. This tends to spread the signals that are represented by the

feature vectors over a larger volume with the result that fewer signatures need to be evaluated by the second-stage correlator. In the preferred embodiment, the inventor has selected five quantization levels and six features as a reasonable compromise. This provides a hyperspace that has 15,625 discrete locations.

Two vector quantizations algorithms were developed. The first partitions each vector according to its statistical distribution so that each region contains the same number of entries (e.g., Figs. 10 and 11). The second performs a similar process but returns two quantized feature vectors (e.g., Figs. 12, 13, and 14). The two vectors represent the most likely region and the nearest neighbor. Histograms were prepared of skew, kurtosis, and bandwidth values using a sample signal library. The histograms for kurtosis will be discussed below with respect to the overlap and non-overlap vector quantization algorithms.

Fig. 12 illustrates the overlap encoded vector quantization of a kurtosis feature. Note that the kurtosis value population is first partitioned so that each region contains one-third of the total number of samples. Thresholds B and D are used to separate regions 1, 3, and 5. Two additional regions are constructed that also each contain one-third of the points. These are centered on thresholds B and D with one-sixth of the total points above and below. Regions 2 and 4 are bounded by thresholds A, C, and E. As indicated by Fig. 13, feature values that are less than threshold A are clearly in region 1 and are encoded as (0,0). Feature values that fall between thresholds A and B may belong in region 1 or 2 and are encoded as (0,1). Those that fall between thresholds B and C may belong in region 2 or 3 and are encoded as (1,2). Values between C and D fall in regions 3 or 4 and are encoded as (2,3). Values between D and E may belong to region 4 or 5 and are encoded as (3,4). Finally, if a value is greater than threshold E, it is clearly in region 5 and is encoded as (4,4). Since each vector utilizes 6 features, the output of the overlap vector quantization process produces a pair of six element vectors as depicted in Figure 14. By permuting the elements of these two vectors, up to 64 new vectors can be formed that specify locations in hyperspace where the signal may be stored. However, each time a (0,0) or a (4,4) code is produced, duplicate vectors are generated. With the sample library used in preliminary investigations, it has been observed that an average of 21.4 distinct vectors are generated by permuting elements of overlapped quantized vectors A and B. These distinct vectors are termed the memory "neighborhood" of a song in hyperspace.

An alternative to the overlapped quantization algorithm is the non-overlapped quantization algorithm depicted in Figs. 10 and 11. Figs. 10 and 11 depict a histogram of kurtosis values generated from the sample library. According to the non-overlap vector quantization algorithm, quantization thresholds are derived by sorting all of the feature values in ascending order and then partitioning this list into five equal sized regions. The data values located at the boundaries are selected as the thresholds. This ensures that each region will contain the same number of samples and the resulting hyperspace will be filled uniformly. Feature vector values falling within any of the regions 1, 2, 3, 4, or 5 are simply encoded with the value, as shown in Fig. 11. This process is workable in generating the hyperspace in memory, although the overlap quantized vector algorithm is preferred. However, the non-overlap quantization algorithm will be used during the process of recognizing the received signal discussed below.

The final process to be discussed with respect to the technical approach is how the quantized vectors are then used to either write into or extract information from a data base containing signal identifications. The present inventor has investigated two approaches to storing and retrieving information that could be applied to the recognition of signal patterns. The first involves storing all of the variations of the expected pattern in hyperspace. This is simply a matter of assigning the song identification code to each location described by the permutation of the two vectors returned by the overlapped vector quantization process. Access involves the computation of a single location that contains a list of patterns that meet the search criteria. This is the most efficient from the point of view of real-time recognition, but requires the storage of the identification code at many locations in the feature hyperspace. An alternate approach is to store each pattern only once in space. When the space is accessed to find a list of candidate songs, all of the variations that are observed in the overlapped quantization process must be searched. The lists that are found at each location must then be merged. While this second approach conserves memory, it slows down the recognition process. A "belt and suspenders" process is also possible that stores all expected variations of the features, and then searches the region for distortions. While this method is probably the most robust, it pays both the time and memory penalties. The first method was selected for further pursuit since real-time performance is important and it is not felt that the limits of memory are being stretched.

There are many methods for mapping the signal identifications into memory, and then retrieving the identifications in accordance with a received signal. The presently preferred embodiment utilizes one possible approach, although the invention is to be construed as covering all obvious equivalents. According to the parameters described above, there are 15,625 discrete locations in the vector hyperspace. Therefore, a table of pointers has been constructed which point to lists containing identification codes of potential pattern matches. Each list has a variable length that is updated when signals are added or removed. Fig. 15 illustrates this access method. When a signal is to be added to the data base, all permutations of the two vectors derived from the overlapped quantization process are computed. A position on the list is then computed as a weighted sum of the vector components using a non-decimal radix, e.g. 5. Each position in this table may point to an empty list or to the location of a list containing the identification codes of signals

that map to this location in hyperspace. These lists are dynamically generated and modified. The first entry in the signal list indicates the number of identification codes that follow. The rest of the list contains the identification codes. Adding a signal identification to the list involves testing to see if the list exists. If not, it must be created. The signal identification is then added, and the list size is set to one. If the list already exists, a check is made to see if the signal identification is already on the list. If it is already on the list, the table is simply exited. If not, the list is expanded by 1, and a new identification code is appended. The first entry in the signal list is also updated to indicate that an additional signal identification code has been added. Removing a signal identification code is a similar process.

The performance of this approach is dependent on the stability of the statistical features chosen. Moments were selected since they are unaffected by speed variations. However, if time offsets are involved, it is quite possible that segments of the song will exhibit different properties that will place it in a different region of the hyperspace. This will result in missed detections. This was investigated by evaluating the feature stability at several time delays and most were found to be stable. However, some changed significantly with time delay. To counter this, statistics are computed over a set of fourteen second intervals within the 28 seconds used by the correlator. At each offset, a set of overlapped locations in the hyperspace is computed, and the song ID code is added to the corresponding lists. If the song is stable, no new entries are generated. However, if the statistics change with time delay, the song is added to all regions in which it might be found to exist.

This concludes a discussion of the technical approach and the overview of the present invention.

Fig. 1 depicts a block diagram of the apparatus according to the present invention, and Fig. 16 depicts a top-level flow chart for the process according to the present invention. In the present invention, the processing structure of Fig. 1 allows simultaneous processing of up to five audio channels. Therefore, up to five broadcast stations may be monitored and their broadcast programs classified. Additional hardware and software modifications could be performed to increase or decrease the number of channels simultaneously monitored.

Antenna 2 receives radio waves bearing audio signals. The antenna apparatus is capable of receiving up to five radio channels simultaneously. The audio signal is received by audio channel receiver 4, and provided to audio preprocessor 6. Audio preprocessor 6 includes filter banks 8, envelope detectors 10, and low pass filters 12, as depicted in Figure 2. Alternatively, the audio preprocessor 6 may be incorporated digitally into the processor 24.

Fig. 1 also depicts analog-to-digital converter 14 which may be used to digitize the audio signal. Multiplexor 16 is used to carry out the multiplexing operations when a plurality of audio channels are being simultaneously monitored. Both A/D converter 14 and multiplexor 16 are coupled to bus 100. Also coupled to bus 100 is an array processor 18. Feature vector operations are carried out in array processor 18 and include the time warping of the second stage feature sequence and the second stage correlation computations.

Processor 24 is also coupled to buss 100 and performs the functions of control, data base management, and all in/out (I/O) management in the first stage classification calculations. Processor 24 may include a microprocessor 26, a memory 28, I/O interfaces 30, a real-time clock 32, reference pattern memory 34, and an off-line memory 36. Preferably, microprocessor 26 may be a Motorola 68020 series microprocessor. Preferably, working memory 28 includes at least 8 Megabytes of memory. Likewise, pattern memory 34 stores both the first stage and second stage reference libraries and preferably is realized by at least a 140 Megabyte hard disk. The off-line memory 36 may be used to change/add/delete reference patterns from the reference pattern libraries in memory 34. Preferably, off-line memory 36 comprises a tape cartridge.

Finally, the processing system may be coupled with such peripherals as a CRT 38, a printer or modem 40, and terminal 42. Such peripherals are coupled to the system through I/O interfaces 30.

Turning now to Figure 2, a coarse spectral analysis will be described. The received audio signal is provided to audio preprocessor 6 where it is divided into a plurality of channels. In the presently preferred embodiment, four channels have been selected. However, greater or fewer channels may be used depending upon the exact type of signal which is to be classified. Each channel includes a bandpass filter 8, each filter having a different value. Each channel also includes a rectifier 10 and a low-pass filter 12. The purpose of the audio preprocessor is to reduce the amount of information processed in the first stage. This provides a long term averaging of the first stage features. Since the purpose of the first stage is to reduce the computations required for recognition, it is desirable to reduce the amount of information processed per unit time. Signal discrimination accuracy is proportional to the time bandwidth product of the feature vector. Therefore, by reducing the feature vector bandwidth while expanding duration, accuracy is maintained while required processing per unit time is decreased. This is true for any process that requires continuous searching for time series events.

In order to accomplish this, the audio input signal depicted in Figure 3 is provided to each of bandpass filters 8. Each bandpass filter outputs a filtered signal, for example, the one depicted in Figure 3. The filtered signals are provided to the rectifiers 10, each of which outputs a waveform shown in Figure 3. Finally, the rectified signals are provided to lowpass filters 12, each of which outputs a lowpass filtered signal, as depicted in Figure 3. By sampling the reduced bandwidth signal, processing time is conserved while simultaneously reducing the sensitivity of the system to speed variations in the audio signal. Therefore, from lowpass filters 12 are provided a plurality of waveforms as depicted in

Figure 4. These waveforms are respectively denoted by $X_1(t)$, $X_2(t)$, $X_3(t)$, and $X_4(t)$. Each of these waveforms is provided to processor 24 which generates the feature sequences according to the waveforms.

Processor 24 thus provides a plurality of feature sequences denoted by $X_{s1}(t)$, $X_{s2}(t)$, $X_{s3}(t)$, and $X_c(t)$ (Fig. 2). Each of these feature sequences is formed as a linear combination of the waveforms $X_1(t)$ through $X_4(t)$. In the present invention, the linear combination is nothing more than a difference between two bands. For example, signal $X_{s1}(t)$ is the linear combination of $X_4(t)$-$X_3(t)$ (hereinafter denoted "band (4-3)"). Likewise, signals for band (3-2) and band (2-1) are produced in processor 24. This step is depicted in Figure 16 at step S110.

However, in some implementations it will be desirable to form sequences that are linear combinations of the four input waveforms and have certain specific characteristics such as orthogonality. Additional preprocessing may be applied at this point to compensate for distortion induced by the broadcaster. An example of such distortion is the compression of dynamic range or expanding it using a logarithmic compression function. Such distortions could be compensated by simulating the dynamic range distortion when creating the data base and/or compensating the received signal for the dynamic range distortion.

The next step is to calculate the statistical moments of each of the band difference signals, as denoted at step S120 of Figure 16. Each band difference signal must be sampled at a plurality of points to generate signal values used to calculate these statistical moments. Figure 5 depicts the sampling of band (4-3). The sampling process is slightly different for generating the reference library, and for recognizing the received signal. When the reference library is generated, the signal is played into the apparatus of Figure 1 at audio channel receiver 4. The second stage correlation process depicted at steps S210 and S270 in Figure 16 is used to choose the most spectrally distinct portion of the signal for use in the reference libraries. Specifically, the processor identifies the most spectrally distinct 28 seconds of the signal for use in generating the first and second stage reference libraries. On the other hand, when a broadcast signal is monitored by the Figure 1 structure, the signal is broken down into a succession of fourteen second segments, and each segment is analyzed and compared with the reference signal libraries.

According to Figure 5, the reference library entry for the target signal is determined by repeatedly sampling each band difference signal with a 128 sampling point window through nine intervals, shifting the window by 16 points for each interval. The signal values determined at the sampling points are then used to calculate the statistical moments skew and kurtosis, as noted at step S120 in Figure 16. Figure 17 depicts the process whereby the statistical moments are calculated. First, the mean of the sampled signal values is obtained according to:

$$\mu = (1/N) \sum_{n} X(n) \qquad \ldots\ldots (1)$$

where N is the number of sampling points, 1<n<N, and X is the value of the signal (e.g. voltage) at the sampling point. Then, the variance of the sampled signal is obtained according to:

$$\text{variance} = (1/N) \sum_{n} (X(n) - \mu)^2 \qquad \ldots\ldots (2)$$

From the variance, the standard deviation is computed as follows:

$$\sigma = \sqrt{\text{variance}} \qquad (3)$$

Next, the statistical moment or moments can be calculated according to the formula:

$$M_k = (1/N) \sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k} \qquad \ldots\ldots (4)$$

While the present invention utilizes two statistical moments, skew and kurtosis, a single statistical moment according to Formula (4) can be calculated, or alternatively one or more higher-order moments may be used. According to the preferred embodiment, the skew is calculated as:

$$s = (1/N) \sum_{n} \frac{(X(n) - \mu)^3}{\sigma^3} \qquad \ldots\ldots (5)$$

And, kurtosis is computed as follows:

$$K = (1/N) \sum_{n} \frac{(X(n) - \mu)^4}{\sigma^4} \qquad \ldots\ldots (6)$$

The above-described calculations are carried out in steps S121-S125, as depicted in Figure 17.

Once the statistical moments have been calculated, the feature vectors can be derived (step S130 in Fig. 16). As depicted in Figure 6, a feature vector is derived for each of the nine intervals during the creation of the reference library. Each vector includes skew and kurtosis values for the three band difference signals at each interval. Thus, vector 1 includes values for $S(4\text{-}3)_1$, $S(3\text{-}2)_1$, $S(2\text{-}1)_1$, $K(4\text{-}3)_1$, $K(3\text{-}2)_1$, and $K(2\text{-}1)_1$. These values are actual signal values and are represented by floating point numbers.

Now, nine feature vectors have been generated for each signal, and each feature vector describes a location in hyperspace. A song identification code will then be placed in each neighborhood in hyperspace, the neighborhood eliminating duplicate values.

The next step is to quantize the feature vectors, as discussed earlier (step S140 in Fig. 16). During the process of creating the reference libraries, the overlapped vector quantization algorithm according to Figures 12-14 is preferably utilized. For example, as depicted in Figure 14, the values of vector 1 are plotted on the graph of Figure 12, and two overlapped quantized vectors A and B are produced. These quantized vector components are represented by a single digit having a value of 0-4. However, in an alternative embodiment, the quantized vectors may be multi-digit, and may comprise more or less than five distinct values.

After the two quantized vectors A and B are obtained, a permutation of these vectors is performed in order to produce all possible variations thereof. Thus, there are 64 possible locations in hyperspace where the signal values from vector 1 could be stored. Since this permutation is performed for each of the nine feature vectors produced, a total of 576 vectors may have to be mapped in hyperspace for each signal. However, according to the experiments conducted by the inventor, about 21 actual locations in hyperspace are produced for each vector.

Mapping each permutated quantized vector into hyperspace is a simple matter of entering a song identification in a particular location in memory (step S150 in Fig. 16). This is performed according to a process depicted in Figure 15, although a wide number of mapping algorithms could be used.

In Figure 15, a weighted sum of the permutated, quantized vector 2A is produced using a non-decimal radix of 5. That is, the vector values are multiplied by different powers of the base 5. Of course, a different radix can be used, or an entirely different method of producing an address could be used. In the example depicted in Figure 15, the weighted sum of the vector components is equal to 11,117. A table entitled Q TABLE is then accessed (step S160 in Fig. 16) with this address and a pointer is provided pointing to a particular song list #1. During the process of creating the data base, the song identification code is now entered on the song list, and the number of songs on the list is incremented by one. This procedure is repeated for each of the permutated quantized vectors produced from the sample signal, and the identification code for the sample is thus written into a plurality of song lists in memory. This completes the creation of the data base with which the signals are recognized.

During signal recognition, the steps described above are generally similar. The broadcast signal is received, band-pass filtered, rectified, lowpass filtered, and linear combinations of the four bands are produced. The sampling step, during recognition, is somewhat different in that the song is continuously sampled at 14 second intervals. Only a single feature vector is created for each interval, utilizing the skew and kurtosis values calculated as per equations (5) and (6).

The single feature vector for each sampled interval is then quantized using the non-overlap quantization algorithm noted above. Thus, each song interval will produce a single quantized vector. A weighted sum of the single quantized vector components is then obtained and the Q TABLE is accessed using the weighted sum. The pointer from the Q TABLE accesses a single song list. The songs on the single song list will be the candidate songs investigated in the second stage classification procedure. Of course, if a single stage classification process is utilized, the song or songs extracted from the single song list will be identified as the broadcast signal.

The second stage classification process is described in detail in the '562 patent at column 13, line 58 through column 15, line 57; and in Figs. 2, 7, 8, 9, and 14. Basically, the song list or lists generated in the first stage classification process are submitted to the second stage correlator together with the signal $X_c(t)$ output in Figure 2. The signal $X_c(t)$ may be linearly time warped, as discussed in the '562 patent. The Fourier transform of all five time warped and un-warped waveforms is calculated and provided as complex spectra which are compared with the second stage reference patterns stored in the second stage library. Samples from the digitized broadcast signals and the reference waveforms are cross multiplied and inverse Fourier transformed to provide a correlation signal (step S211 in Fig. 18). Next, the correlation functions between each second stage reference pattern and the plurality of time-warped (and un-warped) input signals are compared to select the maximum correlation value for the current input signal (step S213 in Fig. 18). The appropriate waveform with the highest correlation value is selected and compared to a threshold value which determines recognition. As soon as a correlation peak value is determined to be above the predetermined threshold, the signal is "recognized" (step S215 in Fig. 18), and the song identification is logged, and may be depicted on CRT 38.

Thus, the above-described system and method provides for an accurate, reliable, compact, yet inexpensive system for classifying signals.

As noted earlier, the apparatus according to the present invention may be used to generate the first and second stage reference libraries. The procedure for generating the first stage library has been described above, and the procedure for generating the second stage library is fully disclosed in the '562 patent, at column 15, line 64 through column

16, line 43.

Therefore, what has been described above is apparatus and method for automatically classifying signals, and preferably broadcast signals. Since the system is microprocessor based, it can be realized in an extremely small and economical package. For example, the existing prototype comprises a commercial FM radio receiver and a small computer including a Sun 3/150 processor, a Sun CPU board using a Motorola 68020 microprocessor, a plurality of memory boards, a Mercury array processor, a preprocessor, a controller for the disk and peripherals, and a CRT. Those with skill in this field will readily understand the significant advancements in signal recognition disclosed in this patent.

While the present invention has been described in connection with what is presently considered to be the most practical and preferred embodiments, it is to be understood that the invention is not limited to the disclosed embodiments. To the contrary, the present invention is intended to cover various modifications and equivalent arrangements included within the scope of the appended claims.

Claims

1. A method of classifying a signal comprising the steps of:

   receiving the signal;
   sampling the signal at a plurality of points to produce a plurality of signal values;
   deriving a multi-value feature vector from said signal values;
   quantizing said feature vector by replacing each feature vector value with a category value determined from a predetermined quantization pattern which non-uniformly categorizes feature vector values; and
   accessing a signal library with the quantized vector to provide a signal identification code corresponding to the received signal.

2. A method according to claim 1 wherein said predetermined quantization pattern is an overlap quantization pattern.

3. A method according to Claim 1, wherein said quantizing step utilizes a plurality of different predetermined quantization patterns to quantize said feature vector.

4. A method according to claim 1 wherein said deriving step includes the steps of:

   calculating two different statistical moments of said plurality of signal values; and
   forming said feature vector using the two calculated statistical moments.

5. A method according to claim 4 wherein a predetermined quantization pattern is provided for each of the two statistical moments.

6. A method according to claim 4 wherein said two different statistical moments comprise skew and kurtosis.

7. A method according to Claim 1 wherein said receiving step includes the step of envelope detecting said received signal.

8. A method according to claim 1 wherein said quantization step includes the steps of:

   replacing said feature vector with a plurality of quantized vectors in accordance with at least one predetermined quantization pattern having overlapping categories; and
   permuting said plurality of quantized vectors to produce a further plurality of permutated quantized vectors.

9. A method according to claim 8 wherein said accessing step comprises the steps of:

   accessing said signal library with said plurality of permuted quantized vectors to reach a plurality of memory locations; and
   writing in each said memory location a signal identification code corresponding to the received signal.

10. A method according to claim 1 wherein said receiving step includes the steps of:

    spectrally analyzing said signal to provide a plurality of analyzed signals having different frequency bands; and

forming linear combinations of said analyzed waveforms;
and wherein said sampling step includes the step of sampling each linear combination to provide a plurality of sampling points for each linear combination;
and wherein said deriving step includes the steps of:
calculating skew and kurtosis values for each linear combination from the corresponding sampling points; and forming said feature vector to include the skew anc kurtosis values from all of said linear combinations.

11. A method preparing a signal identification library useful in identification of broadcast signals, comprising the steps of:

sampling a signal to be broadcast to provide a plurality of analyzed waveforms for each signal to be broadcast;
deriving a plurality of feature vectors from said analyzed waveforms, at least one feature vector for each sampled signal;
quantizing each feature vector, including the substeps of:

establishing a plurality of quantization levels
distributing the quantization levels non-uniformly over a predetermined statistical distribution;
deriving a plurality of quantization thresholds in accordance with the distributed quantization levels; and
replacing each value of each feature vector with a corresponding quantization value determined by said quantization thresholds;: and
storing a value representing each quantized vector in memory as the signal identification for the corresponding sampled signal.

12. A method according to Claim 11 wherein the step of distributing the quantization levels includes the step of distributing the quantization levels in an overlappe manner, and wherein the step of replacing each value of each feature vector includes the further steps of:

replacing each value of each feature vector with a plurality of quantization values determined by the overlapped quantization levels, each feature vector thus being replaced with a plurality of quantized vectors; and
permuting the plurality of quantized vectors of each feature vector to produce a plurality of permuted vectors;
and wherein said step of storing includes the step of storing, for each sampled signal, values corresponding to the plurality of quantized vectors.

13. A method according to Claim 11 wherein said deriving step includes the step of calculating, for each analyzed waveform, two statistical moments, and forming each feature vector to include the two statistical moments.

14. A method according to anyone of claims 1-10, further comprising the steps of:

calculating a statistical moment of said signal according to the formula:

$$(1/N) \sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

where:

N = the number of sampling points;

n = $1 < n < N$

X = the signal value of said signal at a sampling point;

μ = a mean of the signal values;

σ = a standard deviation of the signal values; and

k = an integer greater than 1;

comparing the calculated statistical moment with a library containing a plurality of stored signal identifications;

and
recognizing the received signal as similar to at least one of said stored signal identifications.

15. A method according to Claim 14 wherein said calculating step includes the steps of calculating the skew and kurtosis of said signal, and wherein said comparing step includes the step of forming a feature vector from said skew and said kurtosis.

16. A method according to Claim 14 wherein said receiving step includes the steps of:

bandpass filtering the received signal to provide a plurality of filtered signals:
rectifying said filtered signals;
low pass filtering the rectified signals: and
computing a plurality of linear combinations of the low pass filtered signals.

17. A method according to Claim 16 wherein said sampling step includes the steps of:

sampling a first one of said linear combinations at the plurality of sampling points to produce a first plurality of signal values; and
sampling a second one of said linear combinations at the plurality of sampling points to produce a second plurality of signal values.

18. A method according to Claim 17 wherein said calculating step includes the steps of:

calculating a first skew value and a first kurtosis value from said first plurality of signal values;
calculating a second skew value and a second kurtosis value from said second plurality of signal values; and
deriving a feature vector comprising said first and second skew values and said first and second kurtosis values.

19. A method according to Claim 18 wherein said comparing step includes the steps of:

quantizing said feature vector in accordance with a plurality of predetermined quantization patterns to provide a quantized vector; and
accessing said library with said quantized vector to locate a signal identification corresponding to the quantized vector.

20. A method according to Claim 19 wherein said accessing step includes the steps of:

forming a weighted sum of the values of said quantized vector using a non-decimal radix;
using the weighted sum as an address to access a pointer table to locate a pointer corresponding to said weighted sum; and
using said pointer to locate a signal identification list containing the signal identification corresponding to the quantized vector.

21. A method according to Claim 20 wherein said recognizing step includes the steps of:

correlating a third one of said plurality of linear combinations with a plurality of stored signals which correspond to the signal identifications contained in said signal identification list; and
selecting one stored signal whose correlation with said third linear combination exceeds a predetermined threshold.

22. A method according to Claim 19 wherein said quantizing step includes the steps of:

categorizing each skew value of said feature vector in accordance with a predetermined skew quantization pattern having categories non-uniformly distributed therein; and
categorizing each kurtosis value of said feature vector in accordance with a predetermined kurtosis quantization pattern having categories non-uniformly distributed therein.

23. A method according to any one of claims 11-13, further comprising the steps of:

receiving a reference signal;
sampling said reference signal at a plurality of sampling points to produce a plurality of signal values;
calculating a statistical moment of the received reference signal according to the formula:

$$(1/N) \sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

where:

N = the number of sampling points;

n = 1 < n < N

X = the sampled signal value at a sampling point;

μ = a mean of the sampled signal values;

σ = a standard deviation of the sampled signal values; and

k = an integer greater than 1;

deriving a feature vector from the calculated statistical moment; and
storing the feature vector or a representation thereof in a memory.

24. A method according to Claim 23 wherein said calculating step includes the step of calculating two statistical moments of the received reference signal, and wherein said deriving step includes the step of deriving said feature vector from both of the calculated statistical moments.

25. A method according to Claim 23 wherein said receiving step includes the steps of:

bandpass filtering the received signal to provide a plurality of filtered signals;
rectifying said filtered signals;
low pass filtering the rectified signals: and
computing a plurality of linear combinations of the low pass filtered signals.

26. A method according to Claim 25 wherein said sampling step includes the steps of:

sampling a first portion of one of said linear combinations at a plurality of points to produce a first plurality of signal values: and
sampling a second portion of said one linear combination at a plurality of points to produce a second plurality of signal values.

27. A method according to claim 26 wherein said calculating step includes the steps of:

calculating the skew and kurtosis of said first plurality of signal values to provide a first skew value and a first kurtosis value; and
calculating the skew and kurtosis of said second plurality of signal values to provide a second skew value and a second kurtosis value.

28. A method according to claim 27 wherein said deriving step includes the steps of:

forming a first feature vector including said first skew value and said first kurtosis value; and
forming a second feature vector including said second skew value and said second kurtosis value.

29. A method according to Claim 28, wherein said deriving step further includes the step of quantizing both said feature vectors by replacing the skew and kurtosis values with integers in accordance with predetermined non-uniform quantization patterns to produce first and second quantized vectors.

30. A method according to claim 29 wherein said storing step includes the steps of:

computing a weighted sum of said first vector using a non-decimal radix:
computing a weighted sum of said second vector using said non-decimal radix;
using the weighted sums to address said memory; and
storing in said memory, at the addresses corresponding to the weighted sums, a signal identification code corresponding to said reference signal.

31. A method according to Claim 28 wherein said deriving step further includes the steps of:

producing first and second overlap quantized vectors from said first feature vector in accordance with predetermined non-uniform overlap quantization patterns;
producing third and fourth overlap quantized vectors from said second feature vector in accordance with said predetermined non-uniform overlap quantization patterns;
forming a first plurality of permutations of said first and second overlap quantized vectors; and
forming a second plurality of permutations of said third and fourth overlap quantized vectors.

32. A method according to claim 31 wherein said storing step includes the steps of:

computing a weighted sum of the values of each one of said first and second pluralities of permutations;
accessing said memory at addresses corresponding to said weighted sums; and
storing in said memory, at areas corresponding to said addresses, a signal identification code identifying said reference signal.

33. Apparatus for classifying a signal comprising:

a receiver for receiving the signal;
a memory;
a processor for (a) sampling the signal at a plurality of points to produce a plurality of signal values, (b) deriving a multi-value feature vector from laid signal values, (c) quantizing said feature vector by replacing each feature vector value with a category value determined from a predetermined quantization pattern which non-uniformly categorizes feature vector values, said pattern being stored in said memory, and (d) accessing a signal library with the quantized vector to provide a signal identification code corresponding to the received signal, said library being stored in said memory.

34. Apparatus according to claim 33 wherein said predetermined quantization pattern is an overlap quantization pattern.

35. Apparatus according to Claim 33, wherein said processor utilizes a plurality of different predetermined quantization patterns to quantize said feature vector.

36. Apparatus according to claim 33 wherein said processor (b1) calculates two different statistical moments of said plurality of signal values, and (b2) forms said feature vector using the two calculated statistical moments.

37. Apparatus according to claim 36 wherein a predetermined quantization pattern is provided for each of the two statistical moments.

38. Apparatus according to claim 36 wherein said two different statistical moments comprise skew and kurtosis.

39. Apparatus according to Claim 33 wherein said processor envelope detects said received signal.

40. Apparatus according to claim 33 wherein said processor (c1) replaces said feature vector with a plurality of quantized vectors in accordance with at least one predetermined quantization pattern having overlapping categories, and (c2) permutes said plurality of quantized vectors to produce a further plurality of permutated quantized vectors.

41. Apparatus according to claim 40 wherein said processor (d1) accesses said signal library with said plurality of permutated quantized vectors to reach a plurality of memory locations, and (d2) writes in each said memory location a signal identification code corresponding to the received signal.

42. Apparatus according to claim 33 wherein said receiver (a) spectrally analyzes said signal to provide a plurality of analyzed signals having different frequency bands, and (b) forms linear combinations of said analyzed waveforms:

and wherein said processor samples each linear combination to provide a plurality of sampling points for each linear combination, calculates skew and kurtosis values for each linear combination from the corresponding sampling points, and forms said feature vector to include the skew and kurtosis values from all of said linear combinations.

43. Apparatus for preparing a signal identification library useful in identification of broadcast signals, comprising:

a receiver for receiving a plurality of signals to be broadcast:
a memory;
a processor for (a) sampling the signals to be broadcast to provide a plurality of analyzed waveforms for each signal to be broadcast; (b) deriving a plurality of feature vectors from said analyzed waveforms, at least one feature vector for each sampled signal, and (c) quantizing each feature vector, including the substeps of:

(c1) establishing a plurality of quantization levels;
(c2) distributing the quantization levels non-uniformly over a predetermined statistical distribution;
(c3) deriving a plurality of quantization thresholds in accordance with the distributed quantization levels; and
(c4) replacing each value of each feature vector with a corresponding quantization value determined by said quantization thresholds; and

said processor (d) storing a value representing each quantized vector in said memory as the signal identification for the corresponding sampled signal.

44. Apparatus according to Claim 43 wherein the quantization levels are distributed in an overlapped Banner, and wherein said processor (c4a) replaces each value of each feature vector with a plurality of quantization values determined by the overlapped quantization levels, each feature vector thus being replaced with a plurality of quantized vectors, and (c4b) permutes the plurality of quantized vectors of each feature vector to produce a plurality of permuted vectors;

and wherein said processor (d1) stores in said memory, for each sampled signal, values corresponding to the plurality of quantized vectors.

45. Apparatus according to Claim 43 wherein said processor (b1) calculates, for each analyzed waveform, two statistical moments, and (b2) forms each feature vector to include the two calculated statistical moments.

46. Apparatus according to any one of claims 33-42, wherein said processor is further provided for (e) calculating a statistical moment of said signal according to the formula:

$$(1/N) \sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

where:

N = the number of sampling points;

n = $1 < n < N$

X = the signal value of said signal at a sampling point:

μ = a mean of the signal values;

σ = a standard deviation of the signal values; and

k = an integer greater than 1,

(f) storing a library containing a plurality of stored signal identifications, (g) comparing the calculated statistical moment with the plurality of stored signal identifications in said library, and (h) recognizing the received signal as

similar to one of said stored signal identifications.

47. Apparatus according to Claim 46 wherein said processing means calculates the skew and kurtosis of said signal, and forms a feature vector from said skew and said kurtosis.

48. Apparatus according to Claim 46 wherein said processing means includes:

means for bandpass filtering the received signal to provide a plurality of filtered signals;
means for rectifying said filtered signals;
means for low pass filtering the rectified signals; and
means for computing a plurality of linear combinations of the low pass-filtered signals.

49. Apparatus according to Claim 48 wherein said processing means (a1) samples a first one of said linear combinations at the plurality of sampling points to produce a first plurality of signal values, and (a2) samples a second one of said linear combinations at the plurality of sampling points to produce a second plurality of signal values.

50. Apparatus according to Claim 49 wherein said processing means (e1) calculates a first skew value and a first kurtosis value from said first plurality of signal values, calculates a second skew value and a second kurtosis value from said second plurality of signal values, and (e2) derives a feature vector comprising said first and second skew values and said first and second kurtosis values.

51. Apparatus according to Claim 50 wherein said processing means (g1) quantizes said feature vector in accordance with a plurality of stored predetermined quantization patterns to provide a quantized vector, and (g2) accesses said library with said quantized vector to locate a signal identification corresponding to the quantized vector.

52. Apparatus according to Claim 51 wherein said processing means (g2a) forms a weighted sum of the values of said quantized vector using a non-decimal radix, (g2b) uses the weighted sum as an address to access a pointer table stored in said library to locate a pointer corresponding to said weighted sum and uses said pointer to access said library to locate a signal identification list containing the signal identification corresponding to the quantized vector.

53. Apparatus according to Claim 52 wherein said processing means (h1) correlates a third one of said plurality of linear combinations with a plurality of stored signals which correspond to the signal identifications contained in said signal identification list and (h2) selects one stored signal whose correlation with said third linear combination exceeds a predetermined threshold.

54. Apparatus according to Claim 51 wherein said processing means (g1a) categorizes each skew value of said feature vector in accordance with a stored predetermined skew quantization pattern having categories non-uniformly distributed therein, and (d1b) categorizes each kurtosis value of said feature vector in accordance with a stored predetermined kurtosis quantization pattern having categories non-uniformly distributed therein.

55. Apparatus according to any one of claims 43-45, wherein

said processor is further provided for (e) calculating a statistical moment of the received reference signal according to the formula:

$$(1/N) \sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

where:

N = the number of sampling points;

n = 1 < n < N

X = the sampled signal value at a sampling point;

μ = a mean of the sampled signal values;

σ = a standard deviation of the sampled signal values; and

k = an integer greater than 1,

(f) deriving a feature vector from the calculated statistical moment, and (g) storing the feature vector or a representation thereof in said memory.

56. Apparatus according to Claim 55 wherein said processing means calculates two statistical moments of the received reference signal, and derives said feature vector from both of the calculated statistical moments.

57. Apparatus according to Claim 55 wherein said processing means bandpass filters the received signal to provide a plurality of filtered signals, rectifies said filtered signals, low pass filters the rectified signals, and computes a plurality of linear combinations of the low pass filtered signals.

58. Apparatus according to Claim 57 wherein said processing means (a1) samples a first portion of one of said linear combinations at a plurality of points to produce a first plurality of signal values, and (a2) samples a second portion of said one linear combination at a plurality of points to produce a second plurality of signal values.

59. Apparatus according to claim 58 wherein said processing means (e1) calculates the skew and kurtosis of said first plurality of signal values to provide a first skew value and a first kurtosis value, and (e2) calculates the skew and kurtosis of said second plurality of signal values to provide a second skew value and a second kurtosis value.

60. Apparatus according to claim 59 wherein said processing means (f1) forms a first feature vector including said first skew value and said first kurtosis value, and (f2) forms a second feature vector including said second skew value and said second kurtosis value.

61. Apparatus according to Claim 60, wherein said processing means (f3) quantizes both said feature vectors by replacing the skew and kurtosis values with integers in accordance with predetermined non-uniform quantization patterns stored in said memory to produce first and second quantized vectors.

62. Apparatus according to claim 61 wherein said processing means (g1) computes a weighted sum of said first vector using a non-decimal radix, (g2) computes a weighted sum of said second vector using said non-decimal radix, (g4) uses the weighted sums to address said memory, and (g5) stores in said memory, at addresses corresponding to the weighted sums, a signal identification code corresponding to said reference signal.

63. Apparatus according to Claim 60 wherein said processing means (f3) produces first and second overlap quantized vectors from said first feature vector in accordance with predetermined non-uniform overlap quantization patterns stored in said memory, (f4) produces third and fourth overlap quantized vectors from said second feature vector in accordance with said predetermined non-uniform overlap quantization patterns stored in said memory, (f5) forms a first plurality of permutations of said first and second overlap quantized vectors, and (f6) forms a second plurality of permutations of said third and fourth overlap quantized vectors.

64. Apparatus according to claim 63 wherein said processing means (g1) computes a weighted sum of the values of each one of said first and second pluralities of permutations, (g2) accesses said memory at addresses corresponding to said weighted sums, and (g3) stores in said memory, at areas corresponding to said addresses, a signal identification code identifying said reference signal.

Patentansprüche

1. Verfahren zur Klassifizierung eines Signals, bestehend aus den Schritten:

Empfangen des Signals,
Abtasten des Signals an mehreren Punkten zur Erzeugung mehrerer Signalwerte,
Ableiten eines Multiwert-Merkmalsvektors aus den Signalwerten,
Quantisieren des Merkmalsvektors durch Ersetzen jedes Merkmalsvektorwertes durch einen aus einem vorbestimmten, ungleichartige Merkmalsvektorwerte kategorisierenden Quantisierungsmuster bestimmten Kategoriewert und

Zugreifen auf eine Signalbibliothek mit dem quantisierten Vektor zur Erzeugung eines mit dem empfangenen Signal korrespondierenden Signalidentifikationskodes.

2. Verfahren nach Anspruch 1, wobei das vorbestimmte Quantisierungsmuster ein Überlappungsquantisierungsmuster ist.

3. Verfahren nach Anspruch 1, wobei der Quantisierungsschritt mehrere verschiedene vorbestimmte Quantisierungsmuster zum Quantisieren des Merkmalsvektors verwendet.

4. Verfahren nach Anspruch 1, wobei der Ableitungsschritt die Schritte

Berechnen zweier unterschiedlicher statistischer Momente der mehreren Signalwerte und
Bilden des Merkmalsvektors unter Verwendung der zwei berechneten statistischen Momente

aufweist.

5. Verfahren nach Anspruch 4, wobei für jedes der zwei statistischen Momente ein vorbestimmtes Quantisierungsmuster vorgesehen ist.

6. Verfahren nach Anspruch 4, wobei die zwei verschiedenen statistischen Momente Asymmetrie und Wölbung aufweisen.

7. Verfahren nach Anspruch 1, wobei der Empfangsschritt den Schritt einer Hüllkurvengleichrichtung des empfangenen Signals aufweist.

8. Verfahren nach Anspruch 1, wobei der Quantisierungsschritt die Schritte

Ersetzen des Merkmalsvektors durch mehreren quantisierte Vektoren entsprechend wenigstens einem vorbestimmten Quantisierungsmuster mit Überlappungskategorien und
Permutieren der mehreren quantisierten Vektoren zur Erzeugung mehrerer weiterer permutierter quantisierter Vektoren

aufweist.

9. Verfahren nach Anspruch 8, wobei der Zugreifschritt die Schritte

Zugreifen auf die Signalbibliothek mit den mehreren permutierten quantisierten Vektoren zum Erreichen mehrerer Speicherstellen und
Schreiben in jede Speicherzelle einen mit dem empfangenen Signal korrespondierenden Signalidentifikationskode

aufweist.

10. Verfahren nach Anspruch 1, wobei der Empfangsschritt die Schritte

Spektralanalysieren des Signals zur Erzeugung mehrerer, verschiedene Frequenzbänder aufweisender analysierter Signale und
Bilden von Linearkombinationen der analysierten Wellenformen

aufweist,
wobei der Abtastschritt den Schritt einer Abtastung jeder Linearkombination zur Erzeugung mehrerer Abtastpunkte für jede Linearkombination
aufweist,
und wobei der Ableitungsschritt die Schritte

Berechnen von Asymmetrie- und Wölbungswerten für alle Linearkombinationen aus den korrespondierenden Abtastpunkten und
Bilden des Merkmalsvektors so, daß er Asymmetrie- und Wölbungswerte aller Linearkombinationen enthält,

aufweist.

11. Verfahren zum Erzeugen eines bei der Identifikation von Rundfunksignalen verwendbaren Signalidentifikationsbibliothek, bestehend aus den Schritten:

Abtasten eines durch Rundfunk zu sendenden Signals zur Erzeugung mehrerer analysierter Wellenformen für jedes durch Rundfunk zu sendende Signal,
Ableiten mehrerer Merkmalsvektoren und wenigstens einen Merkmalsvektor pro abgetastetes Signal aus den analysierten Wellenformen,
Quantisieren jedes Merkmalsvektors mit den Unterschritten:
Herstellen mehrerer Quantisierungspegel,
Verteilen der Quantisierungspegel ungleichartig über einer vorbestimmten statistischen Verteilung,
Ableiten mehrerer Quantisierungsschwellenwerte entsprechend den verteilten Quantisierungspegeln,
Ersetzen jedes Wertes jedes Merkmalsvektors durch einen durch die Quantisierungsschwellenwerte bestimmten korrespondierenden Quantisierungswert, und
Speichern jedes einen quantisierten Vektor darstellenden Wertes in einem Speicher als die Signalidentifikation für das korrespondierende abgetastete Signal.

12. Verfahren nach Anspruch 11, wobei der Schritt der Verteilung der Quantisierungspegel den Schritt einer Verteilung der Quantisierungspegel in überlappender Art und Weise aufweist, und wobei der Schritt des Ersetzens jedes Wertes jedes Merkmalsvektors die weiteren Schritte

Ersetzen jedes Wertes jedes Merkmalsvektors durch mehrere durch die überlappten Quantisierungspegel bestimmten Quantisierungswerte, wobei so jeder Merkmalsvektor durch mehrere quantisierte Vektoren ersetzt wird, und
Permutieren der mehreren quantisierten Vektoren jedes Merkmalsvektors zur Erzeugung mehrerer permutierter Vektoren

aufweist,
wobei der Schritt des Speicherns den Schritt eines Speicherns von mit den mehreren quantisierten Vektoren korrespondierenden Werten für jedes abgetastete Signal aufweist.

13. Verfahren nach Anspruch 11, wobei der Ableitungsschritt den Schritt eines Berechnens zweier statistischer Momente für jede analysierte Wellenform und Bilden jedes Merkmalsvektors so, daß er die zwei statistischen Momente enthält, aufweist.

14. Verfahren nach einem der Ansprüche 1 bis 10, mit den Schritten:

Berechnen eines statistischen Moments des Signals entsprechend der Formel

$$(1/N)\sum_{n}\frac{(X(n)-\mu)^{k}}{\sigma^{k}},$$

wobei
N die Zahl der Abtastpunkte,
$1 < n < N$,
X den Signalwert des Signals an einem Abtastpunkt,
μ ein Mittel der Signalwerte,
σ eine Standardabweichung des Signalwertes und
k eine ganze Zahl größer als 1
bedeuten,
Vergleichen des Berechneten statistischen Moments mit einer mehrere gespeicherte Signalidentifikationen enthaltenden Bibliothek und
Erkennen des empfangenen Signals als ähnlich wenigstens einer der gespeicherten Signalidentifikationen.

15. Verfahren nach Anspruch 14, wobei der Berechnungsschritt die Schritte einer Berechnung der Asymmetrie und Wölbung des Signals aufweist, und wobei der Vergleichsschritt den Schritt einer Bildung eines Merkmalsvektors

aus der Asymmetrie und Wölbung aufweist.

16. Verfahren nach Anspruch 14, wobei der Empfangsschritt die Schritte

Bandpaßfiltern des empfangenen Signals zum Erzeugen mehrerer gefilterter Signale,
Gleichrichten der gefilterten Signale,
Tiefpaßfiltern der gleichgerichteten Signale und
Berechnen mehrerer Linearkombinationen der tiefpaßgefilterten Signale

aufweist.

17. Verfahren nach Anspruch 16, wobei der Abtastschritt die Schritte

Abtasten einer ersten der Linearkombinationen an mehreren Abtastpunkten zur Erzeugung einer ersten Anzahl Signalwerte und
Abtasten einer zweiten der Linearkombinationen an mehreren Abtastpunkten zur Erzeugung einer zweiten Anzahl Signalwerte

aufweist.

18. Verfahren nach Anspruch 17, wobei der Berechnungsschritt die Schritte

Berechnen eines ersten Asymmetriewertes und eines ersten Wölbungswertes aus der ersten Anzahl Signalwerte,
Berechnen eines zweiten Asymmetriewertes und eines zweiten Wölbungswertes für die zweite Anzahl Signalwerte und
Ableiten eines den ersten und zweiten Asymmetriewert und ersten und zweiten Wölbungswert aufweisenden Merkmalsvektors

aufweist.

19. Verfahren nach Anspruch 18, wobei der Vergleichsschritt die Schritte

Quantisieren des Merkmalsvektors entsprechend einer Zahl vorbestimmter Quantisierungsmuster zur Erzeugung eines quantisierten Vektors und
Zugreifen auf die Bibliothek mit dem quantisierten Vektor zur Lokalisierung einer mit dem quantisierten Vektor korrespondierenden Signalidentifikation

aufweist.

20. Verfahren nach Anspruch 19, wobei der Zugreifschritt die Schritte

Bilden einer gewichteten Summe der Werte des quantisierten Vektors unter Verwendung einer nichtdezimalen Grundzahl,
Verwenden der gewichteten Summe als eine Adresse zum Zugreifen auf eine Zeigertabelle zum Lokalisieren eines mit der gewichteten Summe korrespondierenden Zeigers und
Verwenden des Zeigers zum Lokalisieren einer die mit dem quantisierten Vektor korrespondierende Signalidentifikation enthaltenden Identifikationsliste

aufweist.

21. Verfahren nach Anspruch 20, wobei der Erkennungsschritt die Schritte

Korrelieren einer dritten der mehreren Linearkombinationen mit mehreren mit den in der Signalidentifikationsliste enthaltenen Signalidentifikationen korrespondierenden gespeicherten Signalen und
Auswählen eines gespeicherten Signals, dessen Korrelation mit der dritten Linearkombination einen vorbestimmten Schwellenwert überschreitet,

aufweist.

22. Verfahren nach Anspruch 19, wobei der Quantisierungsschritt die Schritte

Kategorisieren jedes Asymmetriewertes des Merkmalsvektors entsprechend einem vorbestimmten Asymmetriequantisierungsmuster, in welchem Kategorien ungleichartig verteilt sind, und
Kategorisieren jedes Wölbungswertes des Merkmalsvektors entsprechend einem vorbestimmten Wölbungsquantisierungsmuster, in welchem Kategorien ungleichartig verteilt sind,

aufweist.

23. Verfahren nach einem der Ansprüche 11 bis 13, mit den Schritten:

Empfangen eines Referenzsignals,
Abtasten des Referenzsignals an mehreren Abtastpunkten zur Erzeugung mehrerer Signalwerte,
Berechnen eines statistischen Moments des empfangenen Referenzsignals entsprechend der Formel

$$(1/N)\sum_{n}\frac{(X(n)-\mu)^{k}}{\sigma^{k}},$$

wobei
N die Zahl der Abtastpunkte,
$1 < n < N$,
X den abgetasteten Signalwert an einem Abtastpunkt,
μ ein Mittel der abgetasteten Signalwerte,
σ eine Standardabweichung der abgetasteten Signalwerte und
k eine ganze Zahl größer als 1
bedeuten,
Ableiten eines Merkmalsvektors aus dem berechneten statistischen Moment, und
Speichern des Merkmalsvektors oder einer Darstellung desselben in einem Speicher

aufweist.

24. Verfahren nach Anspruch 23, wobei der Berechnungsschritt den Schritt einer Berechnung zweier statistischer Momente des empfangenen Referenzsignals aufweist, und wobei der Ableitungsschritt den Schritt einer Ableitung des Merkmalsvektors aus den beiden berechneten statistischen Momente aufweist.

25. Verfahren nach Anspruch 23, wobei der Empfangsschritt die Schritte

Bandpaßfiltern des empfangenen Signals zur Erzeugung mehrerer gefilterter Signale,
Gleichrichten der gefilterten Signale,
Tiefpaßfiltern der gleichgerichteten Signale und
Berechnen mehrerer Linearkombinationen der tiefpaßgefilterten Signale

aufweist.

26. Verfahren nach Anspruch 25, wobei der Abtastschritt die Schritte

Abtasten eines ersten Teils einer der Linearkombinationen an mehreren Punkten zur Erzeugung einer ersten Anzahl Signalwerte und
Abtasten eines zweiten Teils der einen Linearkombination an mehreren Punkten zur Erzeugung einer zweiten Anzahl Signalwerte

aufweist.

27. Verfahren nach Anspruch 26, wobei der Berechnungsschritt die Schritte

Berechnen der Asymmetrie und Wölbung der ersten Anzahl Signalwerte zur Erzeugung eines ersten Asymmetriewertes und ersten Wölbungswertes und
Berechnen der Asymmetrie und Wölbung der zweiten Anzahl Signalwerte zur Erzeugung eines zweiten Asymmetriewertes und zweites Wölbungswertes

aufweist.

28. Verfahren nach Anspruch 27, wobei der Ableitungsschritt die Schritte

Bilden eines ersten Merkmalsvektors mit dem ersten Asymmetriewert und ersten Wölbungswert und
Bilden eines zweiten Merkmalsvektors mit dem zweiten Asymmetriewert und zweiten Wölbungswert

aufweist.

29. Verfahren nach Anspruch 28, wobei der Ableitungsschritt den Schritt einer Quantisierung beider Merkmalsvektoren durch Ersetzen der Asymmetrie- und Wölbungswerte durch ganze Zahlen entsprechend vorbestimmter ungleichartiger Quantisierungsmuster zur Erzeugung eines ersten und zweiten quantisierten Vektors aufweist.

30. Verfahren nach Anspruch 29, wobei der Speicherschritt die Schritte

Berechnen einer gewichteten Summe des eine nichtdezimale Grundzahl verwendenden ersten Vektors,
Berechnen einer gewichteten Summe des eine nichtdezimale Grundzahl verwendenden zweiten Vektors,
Verwenden der gewichteten Summen zum Adressieren des Speichers, und
Speichern eines mit dem Referenzsignal korrespondierenden Signalidentifikationskode im Speicher bei den mit den gewichteten Summen korrespondieren Adressen

aufweist.

31. Verfahren nach Anspruch 28, wobei der Ableitungsschritt weiter die Schritte

Erzeugen eines ersten und zweiten überlappungsquantisierten Vektors aus dem ersten Merkmalsvektor entsprechend vorbestimmten ungleichartigen Überlappungsquantisierungsmustern,
Erzeugen eines dritten und vierten überlappungsquantisierten Vektors aus dem zweiten Merkmalsvektor entsprechend den vorbestimmten ungleichartigen Überlappungsquantisierungsmustern,
Bilden einer ersten Anzahl Permutationen des ersten und zweiten überlappungsquantisierten Vektors, und
Bilden einer zweiten Anzahl Permutationen des dritten und vierten überlappungsquantisierten Vektors

aufweist.

32. Verfahren nach Anspruch 31, wobei der Speicherschritt die Schritte

Berechnen einer gewichteten Summe der Werte sowohl der ersten als auch zweiten Anzahl Permutationen,
Zugreifen auf den Speicher bei mit den gewichteten Summen korrespondierenden Adressen, und
Speichern eines das Referenzsignal identifizierenden Signalidentifikationskodes in dem Speicher bei mit den Adressen korrespondierenden Bereichen

aufweist.

33. Vorrichtung zum Klassifizieren eines Signals, bestehend aus:

einem Empfänger zum Empfangen des Signals,
einem Speicher,
einem Prozessor zum a) Abtasten des Signals an mehreren Punkten zur Erzeugung mehrerer Signalwerte, b) Ableiten eines Multiwert-Merkmalsvektors aus den Signalwerten, c) Quantisieren des Merkmalsvektors durch Ersetzen jedes Merkmalsvektorwertes durch einen von einem vorbestimmten, Merkmalsvektorwerte ungleichartig kategorisierenden und im Speicher gespeicherten Quantisierungsmuster bestimmten Kategoriewert und d) Zugreifen auf eine im Speicher gespeicherte Signalbibliothek mit dem quantisierten Vektor zur Erzeugung eines mit dem empfangenen Signal korrespondierenden Signalidentifikationskodes.

34. Vorrichtung nach Anspruch 33, wobei das vorbestimmte Quantisierungsmuster ein Überlappungsquantisierungsmuster ist.

35. Vorrichtung nach Anspruch 33, wobei der Prozessor mehrere verschiedene vorbestimmte Quantisierungsmuster zur Quantisierung des Merkmalsvektors benutzt.

36. Vorrichtung nach Anspruch 33, wobei der Prozessor b1) zwei verschiedene statistische Momente der mehreren Signalwerte berechnet und b2) den Merkmalsvektor unter Verwendung der zwei berechneten statistischen Momente bildet.

37. Vorrichtung nach Anspruch 36, wobei für jedes der zwei statistischen Momente ein vorbestimmtes Quantisierungsmuster vorgesehen ist.

38. Vorrichtung nach Anspruch 36, wobei die zwei verschiedenen statistischen Momente Asymmetrie und Wölbung aufweisen.

39. Vorrichtung nach Anspruch 33, wobei der Prozessor das empfangene Signal hüllkurvengleichrichtet.

40. Vorrichtung nach Anspruch 33, wobei der Prozessor c1) den Merkmalsvektor durch mehrere quantisierte Vektoren entsprechend wenigstens einem vorbestimmten Quantisierungsmuster mit überlappenden Kategorien ersetzt und c2) die mehreren quantisierten Vektoren quantisiert, um eine weitere Anzahl permutierter quantisierter Vektoren zu erzeugen.

41. Vorrichtung nach Anspruch 40, wobei der Prozessor d1) mit den mehreren permutierten quantisierten Vektoren auf die Signalbibliothek zugreift, um mehrere Speicherstellen zu erreichen, und d2) in jede der Speicherstellen einen mit dem empfangenen Signal korrespondierenden Signalidentifikationskode schreibt.

42. Vorrichtung nach Anspruch 33, wobei der Empfänger a) das Signal spektralanalysiert, um mehrere analysierte Signale mit verschiedenen Frequenzbändern zu gewinnen, und b) Linearkombinationen der analysierten Wellenformen bildet,
und wobei der Prozessor jede Signalkombination zur Erzeugung mehrerer Abtastpunkte für jede Linearkombination abtastet, Asymmetrie- und Wölbungswerte für jede Linearkombination aus den korrespondierenden Abtastpunkten berechnet und den Merkmalsvektor derart bildet, daß er Asymmetrie- und Wölbungswerte aus allen Linearkombinationen enthält.

43. Vorrichtung zum Erzeugen einer zur Identifikation von Rundfunksignalen verwendbaren Signalidentifikationsbibliothek, bestehend aus:

einem Empfänger zum Empfangen mehrerer durch Rundfunk zu sendender Signale,
einem Speicher,
einem Prozessor zum a) Abtasten der durch Rundfunk zu sendenden Signale zur Erzeugung mehrerer analysierter Wellenformen für jedes durch Rundfunk zu sendende Signal, b) Ableiten mehrerer Merkmalsvektoren aus den analysierten Wellenformen und wenigstens eines Merkmalsvektors für jedes abgetastete Signal und c) Quantisieren jedes Merkmalsvektors, mit den Unterschritten:

c1) Herstellen mehrerer Quantisierungspegel,
c2) Verteilen der Quantisierungspegel ungleichartig über einer vorbestimmten statistischen Verteilung,
c3) Ableiten mehrerer Quantisierungsschwellenwerte entsprechend den verteilen Quantisierungspegeln, und
c4) Ersetzen jedes Wertes jedes Merkmalsvektors durch einen durch die Quantisierungsschwellenwerte bestimmten korrespondierenden Quantisierungswert, wobei

der Prozessor d) einen jeden quantisierten Vektor im Speicher als die Signalidentifikation für das korrespondierende abgetastete Signal darstellenden Wert speichert.

44. Vorrichtung nach Anspruch 43, wobei die Quantisierungspegel überlapptartig verteilt sind, wobei der Prozessor c4a) jeden Wert jedes Merkmalsvektors durch mehrere durch die überlappten Quantisierungspegel bestimmte Quantisierungswerte ersetzt, wobei jeder Merkmalsvektor auf diese Weise durch mehrere quantisierte Vektoren

ersetzt wird, und c4b) die mehreren quantisierten Vektoren jedes Merkmalsvektors permutiert, um mehrere permutierte Vektoren zu erzeugen, und wobei der Prozessor d1) im Speicher für jedes abgetastete Signal Werte speichert, die mit den mehreren quantisierten Vektoren korrespondieren.

45. Vorrichtung nach Anspruch 43, wobei der Prozessor b1) für jede analysierte Wellenform zwei statistische Momente berechnet und b2) jeden Merkmalsvektor derart bildet, daß er die zwei berechneten statistischen Momente enthält.

46. Vorrichtung nach einem der Ansprüche 33 bis 42, wobei der Prozessor e) zur Berechnung eines statistischen Moments des Signals gemäß der Formel

$$(1/N)\sum_{n}\frac{(X(n)-\mu)^{k}}{\sigma^{k}},$$

wobei

N die Zahl der Abtastpunkte,
1 < n < N,
X den Signalwert des Signals an einem Abtastpunkt,
μ ein Mittel des Signalwertes,
σ eine Standardabweichung des Signalwertes und
k eine ganze Zahl größer als 1

bedeuten,
f) zum Speichern einer mehrere gespeicherte Signalidentifikationen enthaltenden Bibliothek, g) Vergleichen des berechneten statistischen Momentes mit den mehreren gespeicherten Signalidentifikationen in der Bibliothek und h) Erkennen des empfangenen Signals als einem der gespeicherten Signalidentifikationen ähnlich vorgesehen ist.

47. Vorrichtung nach Anspruch 46, wobei die Verarbeitungseinrichtung die Asymmetrie und Wölbung des Signals berechnet und aus der Asymmetrie und Wölbung einen Merkmalsvektor bildet.

48. Vorrichtung nach Anspruch 46, wobei die Verarbeitungseinrichtung

eine Einrichtung zur Bandpaßfilterung des empfangenen Signals zur Erzeugung mehrerer gefilterter Signale,
eine Einrichtung zum Gleichrichten der gefilterten Signale,
eine Einrichtung zum Tiefpaßfiltern der gleichgerichteten Signale und
eine Einrichtung zum Berechnen mehrerer Linearkombinationen der tiefpaßgefilterten Signale

aufweist.

49. Vorrichtung nach Anspruch 48, wobei die Verarbeitungseinrichtung a1) eine erste der Linearkombinationen an mehreren Abtastpunkten abtastet, um eine erste Anzahl Signalwerte zu erzeugen, und a2) eine zweite der Linearkombinationen an mehreren Abtastpunkten abtastet, um eine zweite Anzahl Signalwerte zu erzeugen.

50. Vorrichtung nach Anspruch 49, wobei die Verarbeitungseinrichtung e1) einen ersten Asymmetriewert und einen ersten Wölbungswert aus der ersten Anzahl Signalwerte sowie einen zweiten Asymmetriewert und zweiten Wölbungswert für die zweite Anzahl Signalwerte berechnet und e2) einen den ersten und zweiten Asymmetriewert und ersten und zweiten Wölbungswert aufweisenden Merkmalsvektor ableitet.

51. Vorrichtung nach Anspruch 50, wobei die Verarbeitungseinrichtung g1) den Merkmalsvektor entsprechend einer Anzahl gespeicherter vorbestimmter Quantisierungsmuster quantisiert, um einen quantisierten Vektor zu erzeugen, und g2) mit dem quantisierten Vektor auf die Bibliothek zugreift, um eine mit dem quantisierten Vektor korrespondierende Signalidentifikation zu lokalisieren.

52. Vorrichtung nach Anspruch 51, wobei die Verarbeitungseinrichtung g2) eine gewichtete Summe der Werte des eine nichtdezimale Grundzahl verwendenden quantisierten Vektors bildet, g2b) die gewichtete Summe als eine Adresse zum Zugreifen auf eine in der Bibliothek gespeicherte Zeigertabelle zum Lokalisieren eines mit der gewichteten Summe korrespondierenden Zeigers verwendet und den Zeiger zum Zugreifen auf die Bibliothek zum Lokalisieren einer die mit dem quantisierten Vektor korrespondierende Signalidentifikation enthaltenden Signalidentifikationsliste verwendet.

53. Vorrichtung nach Anspruch 52, wobei die Verarbeitungseinrichtung h1) eine dritte der mehreren Linearkombinationen mit mehreren gespeicherten Signalen korreliert, die mit den in der Signalidentifikationsliste enthaltenen Signalidentifikationen korrespondieren, und h2) ein gespeichertes Signal auswählt, dessen Korrelation mit der dritten Linearkombination einen vorbestimmten Schwellenwert überschreitet.

54. Vorrichtung nach Anspruch 51, wobei die Verarbeitungseinrichtung g1a) jeden Asymmetriewert des Merkmalsvektors entsprechend einem gespeicherten vorbestimmten Asymmetriequantisierungsmuster, in welchem Kategorien ungleichartig verteilt sind, kategorisiert, und d1b) jeden Wölbungswert des Merkmalsvektors entsprechend einem gespeicherten vorbestimmten Wölbungsquantisierungsmuster, in welchem Kategorien ungleichartig verteilt sind, kategorisiert.

55. Vorrichtung nach einem der Ansprüche 43 bis 45, wobei der Prozessor e) zum Berechnen eines statistischen Moments des empfangenen Referenzsignals gemäß der Formel

$$(1/N)\sum_{n}\frac{(X(n)-\mu)^k}{\sigma^k},$$

wobei

N die Zahl der Abtastpunkte,
$1 < n < N$,
X den abgetasteten Signalwert an einem Abtastpunkt,
μ ein Mittel der abgetasteten Signalwerte,
σ eine Standardabweichung der abgetasteten Signal werte und
k eine ganze Zahl größer als 1

bedeuten,
f) Ableiten eines Merkmalsvektors aus dem berechneten statistischen Moment und g) Speichern des Merkmalsvektors oder einer Darstellung dieses Vektors in dem Speicher vorgesehen ist.

56. Vorrichtung nach Anspruch 55, wobei die Verarbeitungseinrichtung zwei statistische Momente des empfangenen Referenzsignals berechnet und den Merkmalsvektor aus beiden berechneten statistischen Momenten ableitet.

57. Vorrichtung nach Anspruch 55, wobei die Verarbeitungseinrichtung das empfangene Signal bandpaßfiltert, um mehrere gefilterte Signale zu erzeugen, die gefilterten Signale gleichrichtet, die gleichgerichteten Signale tiefpaßfiltert und mehrere Linearkombinationen der tiefpaßgefilterten Signale berechnet.

58. Vorrichtung nach Anspruch 57, wobei die Verarbeitungseinrichtung a1) einen ersten Teil einer der Linearkombinationen an mehreren Abtastpunkten abtastet, um eine erste Anzahl Signalwerte zu erzeugen, und a2) einen zweiten Teil der einen Linearkombination an mehreren Abtastpunkten abtastet, um eine zweite Anzahl Signalwerte zu erzeugen.

59. Vorrichtung nach Anspruch 58, wobei die Verarbeitungseinrichtung e1) die Asymmetrie und Wölbung der ersten Anzahl Signalwerte berechnet, um einen ersten Asymmetriewert und einen ersten Wölbungswert zu erzeugen, und e2) die Asymmetrie und Wölbung der zweiten Anzahl Signalwerte berechnet, um einen zweiten Asymmetriewert und zweiten Wölbungswert zu erzeugen.

60. Vorrichtung nach Anspruch 59, wobei die Verarbeitungseinrichtung f1) einen den ersten Asymmetriewert und ersten Wölbungswert enthaltenden ersten Merkmalsvektor bildet und f2) einen den zweiten Asymmetriewert und zweiten Wölbungswert enthaltenden zweiten Merkmalsvektor bildet.

61. Vorrichtung nach Anspruch 60, wobei die Verarbeitungseinrichtung f3) beide Merkmalsvektoren durch Ersetzen der Asymmetrie- und Wölbungswerte durch ganze Zahlen entsprechend vorbestimmter, in dem Speicher gespeicherter ungleichartiger Quantisierungsmuster quantisiert, um den ersten und zweiten quantisierten Vektor zu erzeugen.

62. Vorrichtung nach Anspruch 61, wobei die Verarbeitungseinrichtung g1) eine gewichtete Summe aus dem eine nichtdezimale Grundzahl verwendenden ersten Vektor berechnet, g2) eine gewichtete Summe aus dem eine nichtdezimale Grundzahl verwendenden zweiten Vektor berechnet, g4) die gewichteten Summen zum Adressieren des

Speichers verwendet, und g5) in dem Speicher einen mit dem Referenzsignal korrespondierenden Signalidentifikationskode an einer mit den gewichteten Summen korrespondierenden Adresse speichert.

63. Vorrichtung nach Anspruch 60, wobei die Verabeitungseinrichtung f3) einen ersten und zweiten überlappungsquantisierten Vektor aus dem ersten Merkmalsvektor entsprechend den in dem Speicher gespeicherten vorbestimmten ungleichartigen Überlappungsquantisierungsmustern erzeugt, f4) einen dritten und vierten überlappungsquantisierten Vektor aus dem zweiten Merkmalsvektor entsprechend den in dem Speicher gespeicherten vorbestimmten ungleichartigen Überlappungsquantisierungsmuster erzeugt, f5) eine erste Anzahl Permutationen des ersten und zweiten überlappungsquantisierten Vektors bildet und f6) eine zweite Anzahl Permutationen des dritten und vierten überlappungsquantisierten Vektors bildet.

64. Vorrichtung nach Anspruch 63, wobei die Verarbeitungseinrichtung g1) eine gewichtete Summe der Werte sowohl der ersten als auch zweiten Anzahl Permutationen berechnet, g2) auf den Speicher bei mit den gewichteten Summen korrespondierenden Adressen zugreift und g3) in den Speicher einen das Referenzsignal identifizierenden Signalidentifikationskode bei mit den Adressen korrespondierenden Bereichen speichert.

Revendications

1. Procédé de classification d'un signal comprenant les étapes de :

réception du signal ;
échantillonnage du signal en une pluralité de points pour produire une pluralité de valeurs de signal ;
dérivation d'un vecteur de caractéristique multi-valeurs à partir desdites valeurs de signal ;
quantification dudit vecteur de caractéristique en remplaçant chaque valeur de vecteur de caractéristique par une valeur de catégorie déterminée à partir d'un modèle de quantification prédéterminé qui classe en catégories non uniformément des valeurs de vecteurs de caractéristique ; et
accès à une bibliothèque de signaux par le vecteur quantifié pour fournir un code d'identification du signal correspondant au signal reçu.

2. Procédé selon la revendication 1 dans lequel ledit modèle de quantification prédéterminé est un modèle de quantification de chevauchement.

3. Procédé selon la revendication 1, dans lequel ladite étape de quantification utilise une pluralité de modèles de quantification prédéterminés différents pour quantifier ledit vecteur de caractéristique.

4. Procédé selon la revendication 1 dans lequel ladite étape de dérivation comprend les étapes de :

calcul de deux moments statistiques différents de ladite pluralité de valeurs de signal ; et
formation dudit vecteur de caractéristique en utilisant les deux moments statistiques calculés.

5. Procédé selon la revendication 4 dans lequel un modèle de quantification prédéterminé est fourni pour chacun des deux moments statistiques.

6. Procédé selon la revendication 4 dans lequel lesdits deux moments statistiques différents comprennent l'asymétrie et le kurtosis.

7. Procédé selon la revendication 1 dans lequel ladite étape de réception comprend l'étape de détection d'enveloppe dudit signal reçu.

8. Procédé selon la revendication 1 dans lequel ladite étape de quantification comprend les étapes de :

remplacement dudit vecteur de caractéristique par une pluralité de vecteurs quantifiés selon au moins un modèle de quantification prédéterminé ayant des catégories de chevauchement ; et
permutation de ladite pluralité de vecteurs quantifiés pour produire une autre pluralité de vecteurs quantifiés permutés.

9. Procédé selon la revendication 8 dans lequel ladite étape d'accès comprend les étapes de :

accès à ladite bibliothèque de signaux par ladite pluralité de vecteurs quantifiés permutés pour atteindre une pluralité de positions de mémoire ; et
écriture dans chacune desdites positions de mémoire d'un code d'identification du signal correspondant au signal reçu.

10. Procédé selon la revendication 1 dans lequel ladite étape de réception comprend les étapes de :

analyse de façon spectrale dudit signal pour fournir une pluralité de signaux analysés ayant des bandes de fréquence différentes ; et
formation de combinaisons linéaires desdites formes d'onde analysées ;
et dans lequel ladite étape d'échantillonnage comprend l'étape d'échantillonnage de chaque combinaison linéaire pour fournir une pluralité de points d'échantillonnage pour chaque combinaison linéaire ;
et dans lequel ladite étape de dérivation comprend les étapes de :
calcul des valeurs d'asymétrie et de kurtosis pour chaque combinaison linéaire à partir des points d'échantillonnage correspondants ; et
formation dudit vecteur de caractéristique pour contenir les valeurs d'asymétrie et de kurtosis à partir de toutes lesdites combinaisons linéaires.

11. Procédé de préparation d'une bibliothèque de codes d'identification de signal utile pour identifier des signaux de diffusion, comprenant les étapes de :

échantillonnage d'un signal à diffuser pour fournir une pluralité de formes d'ondes analysées pour chaque signal à diffuser ;
dérivation d'une pluralité de vecteurs de caractéristique à partir desdites formes d'onde analysées, au moins un vecteur de caractéristique pour chaque signal échantillonné ;
quantification de chaque vecteur de caractéristique, comprenant les sous-étapes de :
établissement d'une pluralité de niveaux de quantification
distribution des niveaux de quantification non uniformément en une distribution statistique prédéterminée ;
dérivation d'une pluralité de seuils de quantification selon les niveaux de quantification distribués ; et
remplacement de chaque valeur de chaque vecteur de caractéristique par une valeur de quantification correspondante déterminée par lesdits seuils de quantification ; et
stockage d'une valeur représentant chaque vecteur quantifié dans une mémoire comme code d'identification du signal pour le signal échantillonné correspondant.

12. Procédé selon la revendication 11 dans lequel l'étape de distribution des niveaux de quantification comprend l'étape de distribution des niveaux de quantification d'une manière chevauchante, et dans lequel l'étape de remplacement de chaque valeur de chaque vecteur de caractéristique comprend les autres étapes de :

remplacement de chaque valeur de chaque vecteur de caractéristique par une pluralité de valeurs de quantification déterminées par les niveaux de quantifications se chevauchant, chaque vecteur de caractéristique étant ainsi remplacé par une pluralité de vecteurs quantifiés ; et
permutation de la pluralité de vecteurs quantifiés de chaque vecteur de caractéristique pour produire une pluralité de vecteurs permutés ;
et dans lequel ladite étape de stockage comprend l'étape de stockage, pour chaque signal échantillonné, des valeurs correspondant à la pluralité des vecteurs quantifiés.

13. Procédé selon la revendication 11 dans lequel ladite étape de dérivation comprend l'étape de calcul, pour chaque forme d'onde analysée, de deux moments statistiques, et de formation de chaque vecteur de caractéristique pour comprendre les deux moments statistiques.

14. Procédé selon l'une quelconque des revendications 1 à 10, comprenant en outre les étapes de :

calcul d'un moment statistique dudit signal selon la formule :

$$(1 / N)\sum_{n} \frac{(X(n) - \mu)^{k}}{\sigma^{k}}$$

où :

N = le nombre de points d'échantillonnage ;
$n = 1 < n < N$
X = la valeur de signal dudit signal en un point d'échantillonnage ;
μ = une moyenne des valeurs de signal ;
σ = un écart standard des valeurs de signal ; et
k = un entier supérieur à 1 ;

comparaison du moment statistique calculé avec une bibliothèque contenant une pluralité de codes d'identification de signal stockés ; et
reconnaissance du signal reçu comme similaire au moins à une desdits codes d'identification de signal stockés.

15. Procédé selon la revendication 14 dans lequel ladite étape de calcul comprend les étapes de calcul de l'asymétrie et du kurtosis dudit signal, et dans lequel ladite étape de comparaison comprend l'étape de formation d'un vecteur de caractéristique à partir de ladite asymétrie et dudit kurtosis.

16. Procédé selon la revendication 14 dans lequel ladite étape de réception comprend les étapes de :

filtrage passe-bande du signal reçu pour fournir une pluralité de signaux filtrés ;
redressement desdits signaux filtrés ;
filtrage passe-bas des signaux redressés ; et
calcul d'une pluralité de combinaisons linéaires des signaux filtrés passe-bas.

17. Procédé selon la revendication 16 dans lequel ladite étape d'échantillonnage comprend les étapes de :

échantillonnage d'une première desdites combinaisons linéaires en une pluralité de points d'échantillonnage pour produire une première pluralité de valeurs de signal ; et
échantillonnage d'une seconde desdites combinaisons linéaires en une pluralité de points d'échantillonnage pour produire une seconde pluralité de valeurs de signal.

18. Procédé selon la revendication 17 dans lequel ladite étape de calcul comprend les étapes de :

calcul d'une première valeur d'asymétrie et d'une première valeur de kurtosis à partir de ladite première pluralité de valeurs de signal ;
calcul d'une seconde valeur d'asymétrie et d'une seconde valeur de kurtosis à partir de ladite seconde pluralité de valeurs de signal ; et
dérivation d'un vecteur de caractéristique comprenant lesdites première et seconde valeurs d'asymétrie et lesdites première et seconde valeurs de kurtosis.

19. Procédé selon la revendication 18 dans lequel ladite étape de comparaison comprend les étapes de :

quantification desdits vecteurs de caractéristique selon une pluralité de modèles de quantification prédéterminés pour fournir un vecteur quantifié ; et
accès à ladite bibliothèque par ledit vecteur quantifié pour localiser un code d'identification du signal correspondant au vecteur quantifié.

20. Procédé selon la revendication 19 dans lequel ladite étape d'accès comprend les étapes de :

formation d'une somme pondérée des valeurs dudit vecteur quantifié utilisant une base non décimale ;
utilisation de la somme pondérée comme adresse pour accéder à une table de pointeur pour positionner un pointeur correspondant à ladite somme pondérée ; et
utilisation dudit pointeur pour localiser une liste de codes d'identification du signal contenant le code d'identification du signal correspondant au vecteur quantifié.

21. Procédé selon la revendication 20 dans lequel ladite étape de reconnaissance comprend les étapes de :

corrélation d'une troisième combinaison linéaire de ladite pluralité de combinaisons linéaires avec une pluralité de signaux stockés qui correspondent aux codes d'identification de signal contenus dans ladite liste de codes d'identification de signal ; et
sélection d'un signal stocké dont la corrélation avec ladite troisième combinaison linéaire dépasse un seuil

prédéterminé.

22. Procédé selon la revendication 19 dans lequel ladite étape de quantification comprend les étapes de :

classification en catégories de chaque valeur d'asymétrie dudit vecteur de caractéristique selon un modèle de quantification d'asymétrie prédéterminé y ayant des catégories non uniformément distribuées ; et
classification en catégories de chaque valeur de kurtosis dudit vecteur de caractéristique selon un modèle de quantification de kurtosis prédéterminé y ayant des catégories distribuées non uniformément.

23. Procédé selon l'une quelconque des revendications 11 à 13, comprenant en outre les étapes de :

réception d'un signal de référence ;
échantillonnage dudit signal de référence en une pluralité de points d'échantillonnage pour produire une pluralité de valeurs de signal ;
calcul d'un moment statistique du signal de référence reçu selon la formule :

$$(1 / N)\sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

où :

N = le nombre de points d'échantillonnage ;
n = 1 < n < N
X = la valeur de signal échantillonné en un point d'échantillonnage ;
μ = une moyenne des valeurs du signal échantillonné ;
σ = un écart standard des valeurs du signal échantillonné ; et
k = un entier supérieur à 1 ;

dérivation d'un vecteur de caractéristique à partir d'un moment statistique calculé ; et
stockage du vecteur de caractéristique ou d'une représentation de celui-ci dans une mémoire.

24. Procédé selon la revendication 23 dans lequel ladite étape de calcul comprend l'étape de calcul de deux moments statistiques du signal de référence reçu, et dans lequel ladite étape de dérivation comprend l'étape de dérivation dudit vecteur de caractéristique à partir des deux moments statistiques calculés.

25. Procédé selon la revendication 23 dans lequel ladite étape de réception comprend les étapes de :

filtrage passe-bande du signal reçu pour fournir une pluralité de signaux filtrés ;
redressement desdits signaux filtrés;
filtrage passe-bas des signaux redressés ; et
calcul d'une pluralité de combinaisons linéaires des signaux filtrés passe-bas.

26. Procédé selon la revendication 25 dans lequel ladite étape d'échantillonnage comprend les étapes de :

échantillonnage d'une première partie d'une desdites combinaisons linéaires en une pluralité de points pour produire une première pluralité de valeurs de signal ; et
échantillonnage d'une seconde partie d'une desdites combinaisons linéaires en une pluralité de points pour produire une seconde pluralité de valeurs de signal.

27. Procédé selon la revendication 26 dans lequel ladite étape de calcul comprend les étapes de :

calcul de l'asymétrie et du kurtosis de ladite première pluralité de valeurs de signal pour fournir une première valeur d'asymétrie et une première valeur de kurtosis ; et
calcul de l'asymétrie et du kurtosis de ladite seconde pluralité de valeurs de signal pour fournir une seconde valeur d'asymétrie et une seconde valeur de kurtosis.

28. Procédé selon la revendication 27 dans lequel ladite étape de dérivation comprend les étapes de :

formation d'un premier vecteur de caractéristique comprenant ladite première valeur d'asymétrie et ladite première valeur de kurtosis ; et

formation d'un second vecteur de caractéristique comprenant ladite seconde valeur d'asymétrie et ladite seconde valeur de kurtosis.

29. Procédé selon la revendication 28, dans lequel ladite étape de dérivation comprend en outre l'étape de quantification desdits deux vecteurs de caractéristique en remplaçant les valeurs d'asymétrie et de kurtosis par des entiers selon les modèles de quantification non uniforme prédéterminés pour produire les premier et second vecteurs quantifiés.

30. Procédé selon la revendication 29 dans lequel ladite étape de stockage comprend les étapes de :

calcul d'une somme pondérée dudit premier vecteur en utilisant une base non décimale ;
calcul d'une somme pondérée dudit second vecteur en utilisant ladite base non décimale ;
utilisation des sommes pondérées pour adresser ladite mémoire, et
stockage dans ladite mémoire, aux adresses correspondant aux sommes pondérées, d'un code d'identification de signal correspondant audit signal de référence.

31. Procédé selon la revendication 28 dans lequel ladite étape de dérivation comprend en outre les étapes de :

production des premier et second vecteurs quantifiés de chevauchement à partir dudit premier vecteur de caractéristique selon des modèles de quantification de chevauchement non uniformes prédéterminés ;
production des troisième et quatrièmes vecteurs quantifiés de chevauchement à partir dudit second vecteur de caractéristique selon lesdits modèles de quantification de chevauchement non uniforme prédéterminés ;
formation d'une première pluralité de permutations desdits premier et second vecteurs quantifiés de chevauchement; et
formation d'une seconde pluralité de permutations desdits troisième et quatrième vecteurs quantifiés de chevauchement.

32. Procédé selon la revendication 31 dans lequel ladite étape de stockage comprend les étapes de :

calcul d'une somme pondérée des valeurs de chacune desdites première et seconde pluralités de permutations ;
accès à ladite mémoire aux adresses correspondant auxdites sommes pondérées ; et
stockage dans ladite mémoire, à des zones correspondant auxdites adresses, d'un code d'identification de signal identifiant ledit signal de référence.

33. Appareil pour classifier un signal comprenant :

un récepteur pour recevoir le signal ;
une mémoire ;
un processeur pour (a) échantillonner le signal en une pluralité de points pour produire une pluralité de valeurs de signal, (b) dériver un vecteur de caractéristique multi-valeurs à partir desdites valeurs de signal, (c) quantifier ledit vecteur de caractéristique en remplaçant chaque valeur de vecteur de caractéristique par une valeur de catégorie déterminée à partir d'un modèle de quantification prédéterminé qui classe en catégories non uniformément des valeurs de vecteur de caractéristique, ledit modèle étant stocké dans ladite mémoire, et (d) accéder à une bibliothèque de signaux par le vecteur quantifié pour fournir un code d'identification de signal correspondant au signal reçu, ladite bibliothèque étant stockée dans ladite mémoire.

34. Appareil selon la revendication 33 dans lequel ledit modèle de quantification prédéterminé est un modèle de quantification de chevauchement.

35. Appareil selon la revendication 33, dans lequel ledit processeur utilise une pluralité de modèles de quantification prédéterminés différents pour quantifier ledit vecteur de caractéristique.

36. Appareil selon la revendication 33 dans lequel ledit processeur (b1) calcule deux moments statistiques différents de ladite pluralité de valeurs de signal, et (b2) forme ledit vecteur de caractéristique utilisant les deux moments statistiques calculés.

37. Appareil selon la revendication 36 dans lequel un modèle de quantification prédéterminé est fourni pour chacun

des deux moments statistiques.

38. Appareil selon la revendication 36 dans lequel lesdits deux moments statistiques différents comprennent l'asymétrie et le kurtosis.

39. Appareil selon la revendication 33 dans lequel ladite enveloppe de traitement détecte ledit signal reçu.

40. Appareil selon la revendication 33 dans lequel ledit processeur (c1) remplace ledit vecteur de caractéristique par une pluralité de vecteurs quantifiés selon au moins un modèle de quantification prédéterminé ayant des catégories de chevauchement, et (c2) permute ladite pluralité de vecteurs quantifiés pour produire une autre pluralité de vecteurs quantifiés permutés.

41. Appareil selon la revendication 40 dans lequel ledit processeur (d1) accède à ladite bibliothèque de signaux par ladite pluralité de vecteurs quantifiés permutés pour atteindre une pluralité de positions de mémoire, et (d2) écrit dans chacune desdites positions de mémoire un code d'identification de signal correspondant au signal reçu.

42. Appareil selon la revendication 33 dans lequel ledit récepteur (a) analyse spectralement ledit signal pour fournir une pluralité de signaux analysés ayant différentes bandes de fréquence, et (b) forme des combinaisons linéaires desdites formes d'onde analysées ;

et dans lequel ledit processeur échantillonne chaque combinaison linéaire pour fournir une pluralité de points d'échantillonnage pour chaque combinaison linéaire, calcule des valeurs d'asymétrie et de kurtosis pour chaque combinaison linéaire à partir des points d'échantillonnage correspondants, et forme ledit vecteur de caractéristique pour inclure les valeurs d'asymétrie et de kurtosis à partir de toutes lesdites combinaisons linéaires.

43. Appareil pour préparer une bibliothèque de codes d'identification de signal utile pour identifier des signaux de diffusion, comprenant :

un récepteur pour recevoir une pluralité de signaux à diffuser ;
une mémoire ;
un processeur pour (a) échantillonner les signaux à diffuser pour fournir une pluralité de formes d'onde analysées pour chaque signal à diffuser ; (b) dériver une pluralité de vecteurs de caractéristique à partir desdites formes d'onde analysées, au moins un vecteur de caractéristique pour chaque signal échantillonné, et (c) quantification de chaque vecteur de caractéristique, comprenant les sous-étapes de :

(c1) établissement d'une pluralité de niveaux de quantification ;
(c2) distribution des niveaux de quantification non uniformément en une distribution statistique prédéterminée ;
(c3) dérivation d'une pluralité de seuils de quantification selon les niveaux de quantification distribués ; et
(c4) remplacement de chaque valeur de chaque vecteur de caractéristique par une valeur de quantification correspondante déterminée par lesdits seuils de quantification ; et

ledit processeur (d) stockant une valeur représentant chaque vecteur quantifié dans ladite mémoire comme code d'identification du signal pour le signal échantillonné correspondant.

44. Appareil selon la revendication 43 dans lequel les niveaux de quantification sont distribués d'une manière chevauchante, et où ledit processeur (c4a) remplace chaque valeur de chaque vecteur de caractéristique par une pluralité de valeurs de quantification déterminées par les niveaux de quantification se chevauchant, chaque vecteur de caractéristique étant ainsi remplacé par une pluralité de vecteurs quantifiés, et (c4b) permute la pluralité de vecteurs quantifiés de chaque vecteur de caractéristique pour produire une pluralité de vecteurs permutés ;

et dans lequel ledit processeur (d1) stocke dans ladite mémoire, pour chaque signal échantillonné, des valeurs correspondant à la pluralité de vecteurs quantifiés.

45. Appareil selon a revendication 43 dans lequel ledit processeur (b1) calcule, pour chaque forme d'onde analysée, deux moments statistiques, et (b2) forme chaque vecteur de caractéristique pour inclure les deux moments statistiques calculés.

46. Appareil selon l'une quelconque des revendications 33 à 42, dans lequel ledit processeur est de plus prévu pour (e) calculer un moment statistique dudit signal selon la formule :

$$(1 / N)\sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

où :

N = le nombre de points d'échantillonnage ;
$n = 1 < n < N$
X = la valeur de signal dudit signal en un point d'échantillonnage ;
μ = une moyenne des valeurs du signal ;
σ = un écart standard des valeurs du signal ; et
k = un entier supérieur à 1 ;

(f) stockage d'une bibliothèque contenant une pluralité de codes d'identification de signal stockés, (g) comparaison du moment statistique calculé avec la pluralité de codes d'identification de signal stockés dans ladite bibliothèque, et (h) reconnaissance du signal reçu comme similaire à celui desdits codes d'identifications de signaux stockés.

47. Appareil selon la revendication 46 dans lequel ledit dispositif de traitement calcule l'asymétrie et le kurtosis dudit signal, et forme un vecteur de caractéristique à partir de ladite asymétrie et dudit kurtosis.

48. Appareil selon la revendication 46 dans lequel ledit dispositif de traitement comprend :

un dispositif pour filtrer en passe-bande le signal reçu pour fournir une pluralité de signaux filtrés ;
un dispositif pour redresser lesdits signaux filtrés ;
un dispositif pour filtrer en passe-bas les signaux redressés ; et
un dispositif pour calculer une pluralité de combinaisons linéaires Ces signaux filtrés en passe-bas.

49. Appareil selon la revendication 48 dans lequel ledit dispositif de traitement (a1) échantillonne une première desdites combinaisons linéaires en une pluralité de points d'échantillonnage pour produire une première pluralité de valeurs de signal, et (a2) échantillonne une seconde desdites combinaisons linéaires en une pluralité de points d'échantillonnage pour produire une seconde pluralité de valeurs de signal.

50. Appareil selon la revendication 49 dans lequel ledit dispositif de traitement (e1) calcule une première valeur d'asymétrie et une première valeur de kurtosis à partir de ladite première pluralité de valeurs de signal, calcule une seconde valeur d'asymétrie et une seconde valeur de kurtosis à partir de ladite seconde pluralité de valeurs de signal et, (c2) dérive un vecteur de caractéristique comprenant lesdites première et seconde valeurs d'asymétrie et lesdites première et seconde valeurs de kurtosis.

51. Appareil selon la revendication 50 dans lequel ledit dispositif de traitement (g1) quantifie lesdits vecteurs de caractéristique selon une pluralité de modèles de quantification prédéterminés stockés pour fournir un vecteur quantifié, et (g2) accède à ladite bibliothèque par ledit vecteur quantifié pour localiser un code d'identification du signal correspondant au vecteur quantifié.

52. Appareil selon la revendication 51 dans lequel ledit dispositif de traitement (g2a) forme une somme pondérée des valeurs dudit vecteur quantifié utilisant une base non décimale, (g2b) utilise la somme pondérée comme une adresse pour accéder à une table de pointeur stockée dans ladite bibliothèque pour positionner un pointeur correspondant à ladite somme pondérée et utilise ledit pointeur pour accéder à ladite bibliothèque pour localiser une liste de codes d'identification de signal contenant le code d'identification du signal correspondant au vecteur quantifié.

53. Appareil selon la revendication 52 dans lequel ledit dispositif de traitement (h1) corrèle une troisième combinaison linéaire de ladite pluralité de combinaisons linéaires avec une pluralité de signaux stockés qui correspondent aux codes d'identification de signal contenus dans ladite liste de codes d'identification de signal et (h2) sélectionne un signal stocké dont la corrélation avec ladite troisième combinaison linéaire dépasse un seuil prédéterminé.

54. Appareil selon la revendication 51 dans lequel ledit dispositif de traitement (g1a) classe en catégories chaque valeur d'asymétrie dudit vecteur de caractéristique selon un modèle de quantification d'asymétrie prédéterminé stocké y ayant des catégories distribuées non uniformément, et (d1b) classe en catégorie chaque valeur de kurtosis dudit vecteur de caractéristique selon un modèle de quantification de kurtosis prédéterminé stocké y ayant des

catégories non uniformément distribuées.

55. Appareil selon l'une quelconque des revendications 43 à 45, dans lequel ledit processeur est en outre prévu pour (e) calculer un moment statistique du signal de référence reçu selon la formule :

$$(1 / N)\sum_{n} \frac{(X(n) - \mu)^k}{\sigma^k}$$

où :

N = le nombre de points d'échantillonnage ;
n = 1 < n < N
X = la valeur du signal échantillonné en un point d'échantillonnage ;
μ = une moyenne des valeurs du signal échantillonné;
σ = un écart standard des valeurs du signal échantillonné ; et
k = un entier supérieur à 1 ;

(f) dérivation d'un vecteur de caractéristique à partir du moment statistique calculé, et (g) stockage du vecteur de caractéristique ou une représentation de celui-ci dans ladite mémoire.

56. Appareil selon la revendication 55 dans lequel ledit dispositif de traitement calcule deux moments statistiques du signal de référence reçu, et dérive ledit vecteur de caractéristique à partir des deux moments statistiques calculés.

57. Appareil selon la revendication 55 dans lequel ledit dispositif de traitement filtre en passe-bande le signal reçu pour fournir une pluralité de signaux filtrés, redresse lesdits signaux filtrés, filtre en passe-bas les signaux redressés, et calcule une pluralité de combinaisons linéaires des signaux filtrés en passe-bas.

58. Appareil selon la revendication 57 dans lequel ledit dispositif de traitement (a1) échantillonne une première partie d'une desdites combinaisons linéaires en une pluralité de points pour produire une première pluralité de valeurs de signal, et (a2) échantillonne une seconde partie de ladite combinaison linéaire en une pluralité de points pour produire une seconde pluralité de valeurs de signal.

59. Appareil selon la revendication 58 dans lequel ledit dispositif de traitement (e1) calcule l'asymétrie et le kurtosis de ladite première pluralité de valeurs de signal pour fournir une première valeur d'asymétrie et une première valeur de kurtosis, et (e2) calcule l'asymétrie et le kurtosis de ladite seconde pluralité de valeurs du signal pour fournir une seconde valeur d'asymétrie et une seconde valeur de kurtosis.

60. Appareil selon la revendication 59 dans lequel ledit dispositif de traitement (f1) forme un premier vecteur de caractéristique comprenant ladite première valeur d'asymétrie et ladite première valeur de kurtosis, et (f2) forme un second vecteur de caractéristique comprenant ladite seconde valeur d'asymétrie et ladite seconde valeur de kurtosis.

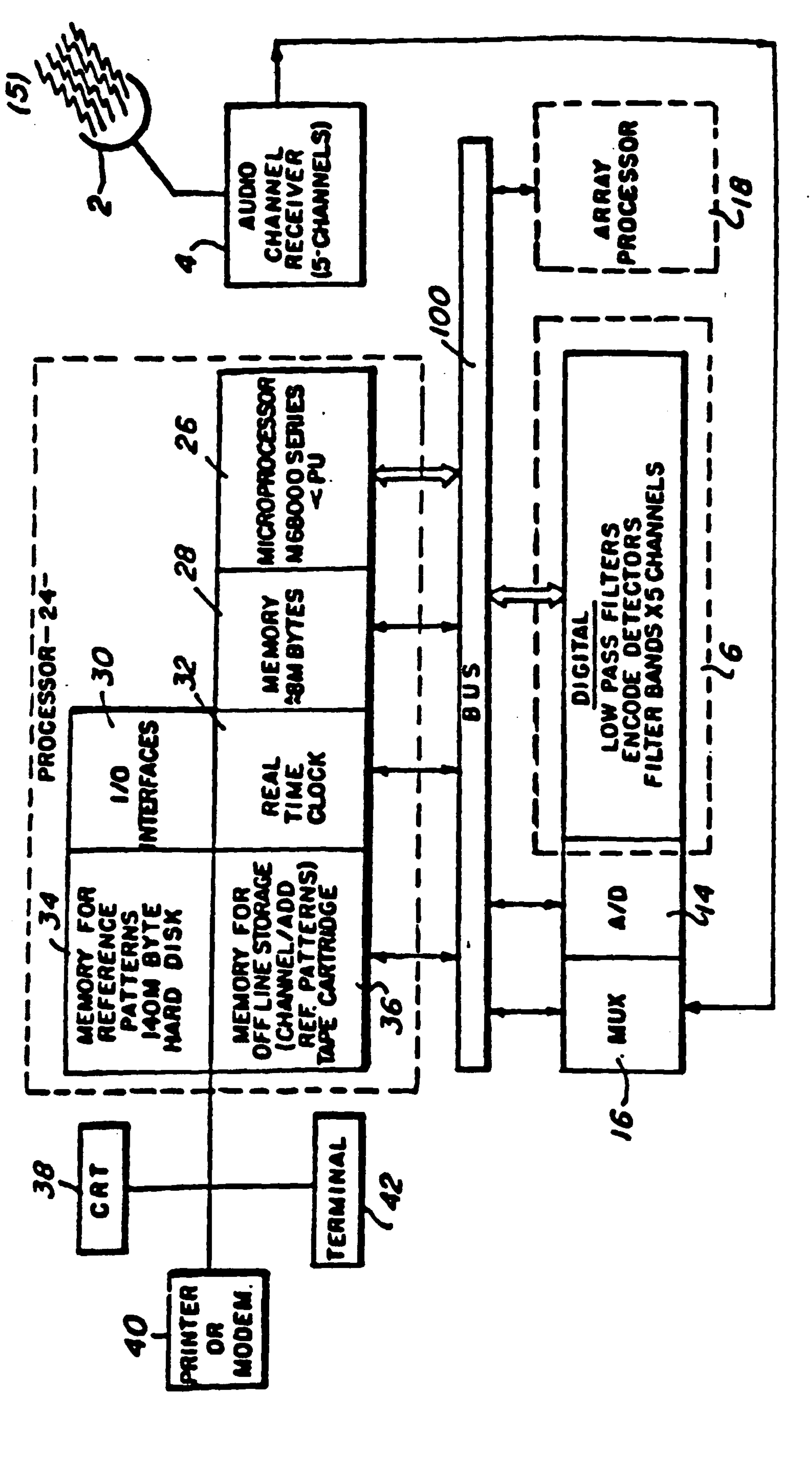
61. Appareil selon la revendication 60, dans lequel ledit dispositif de traitement (f3) quantifie lesdits deux vecteurs de caractéristique en remplaçant les valeurs d'asymétrie et de kurtosis par des entiers selon des modèles de quantification non uniformes prédéterminés stockés dans ladite mémoire pour produire des premier et second vecteurs quantifiés.

62. Appareil selon la revendication 61 dans lequel ledit dispositif de traitement (g1) calcule une somme pondérée dudit premier vecteur utilisant une base non décimale, (g2) calcule une somme pondérée dudit second vecteur utilisant ladite base non décimale, (g4) utilise les sommes pondérées pour adresser ladite mémoire, et (g5) stocke dans ladite mémoire, aux adresses correspondant aux sommes pondérées, un code d'identification du signal correspondant audit signal de référence.

63. Appareil selon la revendication 60 dans lequel ledit dispositif de traitement (f3) produit des premier et second vecteurs quantifiés de chevauchement à partir dudit premier vecteur de caractéristique selon des modèles de quantification de chevauchement non uniformes prédéterminés stockés dans ladite mémoire, (f4) produit des troisième et quatrième vecteurs quantifiés de chevauchement à partir dudit second vecteur de caractéristique selon lesdits modèles de quantification de chevauchement non uniformes stockés dans ladite mémoire, (f5) forme une première pluralité de permutations desdits premier et second vecteurs quantifiés de chevauchement, et (f6) forme une seconde pluralité de permutations desdits troisième et quatrième vecteurs quantifiés de chevauchement.

64. Appareil selon la revendication 63 dans lequel ledit dispositif de traitement (g1) calcule une somme pondérée des valeurs de chacune desdites première et seconde pluralités de permutations, (g2) accède à ladite mémoire aux adresses correspondant auxdites sommes pondérées, et (g3) stocke dans ladite mémoire, à des zones correspondant auxdites adresses, un code d'identification de signal identifiant ledit signal de référence.

FIG. 1


(5)
2
4
AUDIO CHANNEL RECEIVER (5-CHANNELS)
ARRAY PROCESSOR
18
100
BUS
PROCESSOR-24-
30
I/O INTERFACES
32
REAL TIME CLOCK
28
MEMORY 2M BYTES
26
MICROPROCESSOR M68000 SERIES <PU
34
MEMORY FOR REFERENCE PATTERNS 140M BYTE HARD DISK
MEMORY FOR OFF LINE STORAGE (CHANNEL/ADD REF. PATTERNS) TAPE CARTRIDGE
36
DIGITAL
LOW PASS FILTERS
ENCODE DETECTORS
FILTER BANDS X5 CHANNELS
6
A/D
44
MUX
16
38
CRT
TERMINAL
42
40
PRINTER OR MODEM

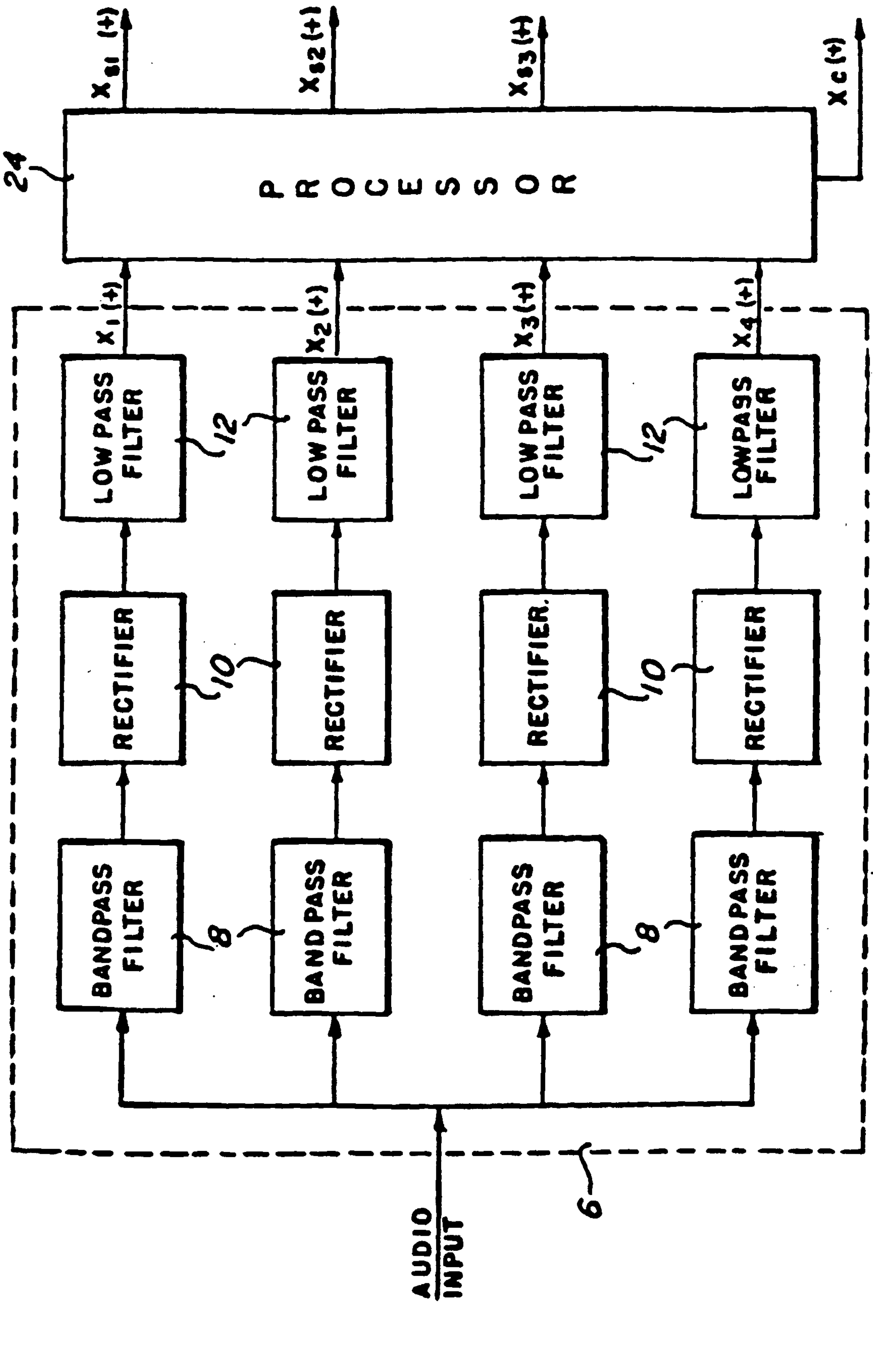

24
PROCESSOR
$X_{S1}(t)$
$X_{S2}(t)$
$X_{S3}(t)$
$X_C(t)$
$X_1(t)$
$X_2(t)$
$X_3(t)$
$X_4(t)$
LOW PASS FILTER
LOW PASS FILTER
LOW PASS FILTER
LOWPASS FILTER
12
12
RECTIFIER
RECTIFIER
RECTIFIER.
RECTIFIER
10
10
BANDPASS FILTER
BAND PASS FILTER
BANDPASS FILTER
BANDPASS FILTER
8
8
6
AUDIO INPUT

FIG. 2

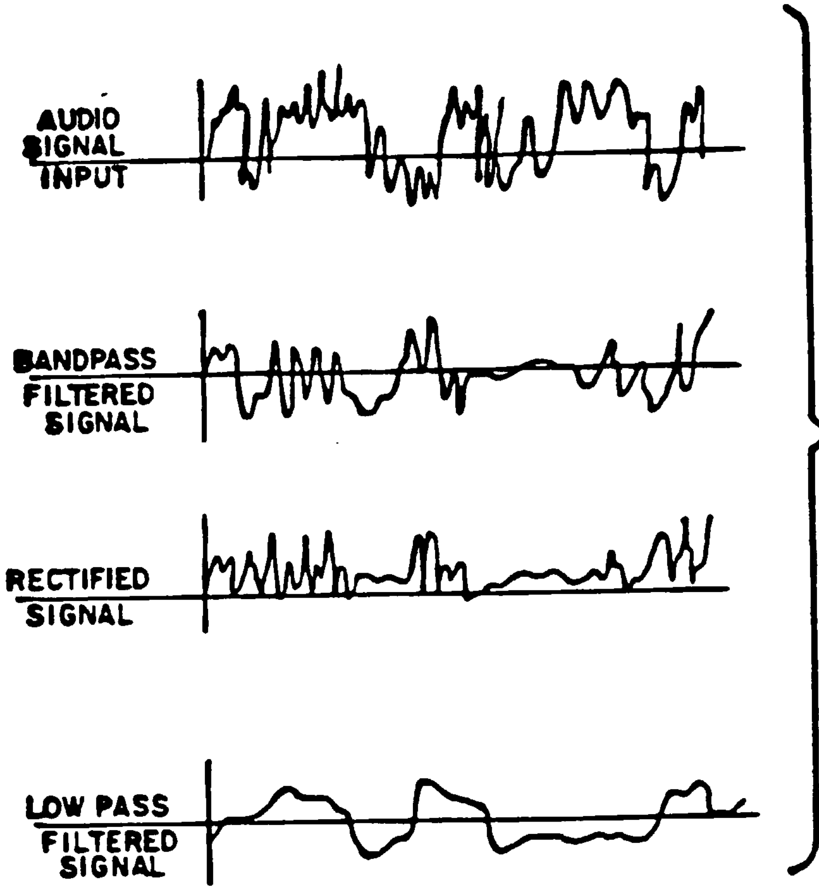

AUDIO SIGNAL INPUT
BANDPASS FILTERED SIGNAL
RECTIFIED SIGNAL
LOW PASS FILTERED SIGNAL

FIG. 3

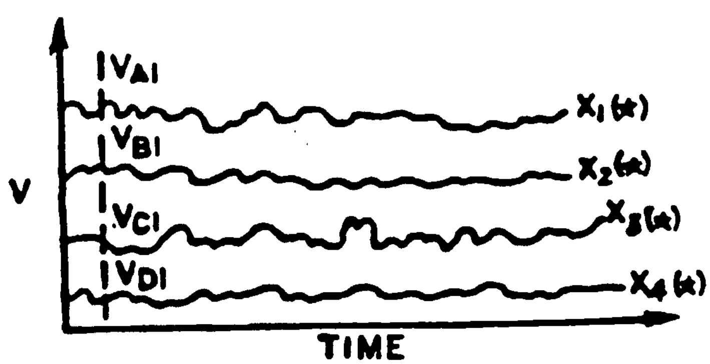

V
V_{A1}
V_{B1}
V_{C1}
V_{D1}
$X_1(k)$
$X_2(k)$
$X_3(k)$
$X_4(k)$
TIME

FIG. 4

FIG. 5

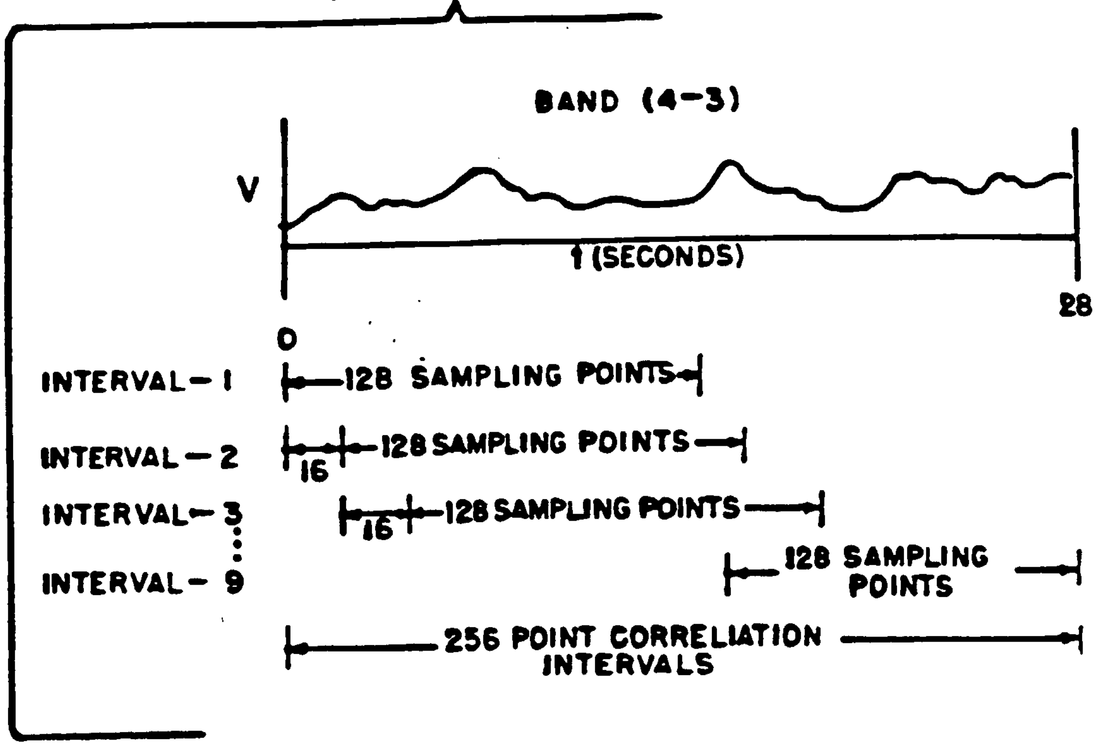

BAND (4-3)
V
t (SECONDS)
0
28
INTERVAL-1
128 SAMPLING POINTS
INTERVAL-2
16
128 SAMPLING POINTS
INTERVAL-3
16
128 SAMPLING POINTS
INTERVAL-9
128 SAMPLING POINTS
256 POINT CORRELATION INTERVALS

FIG. 6

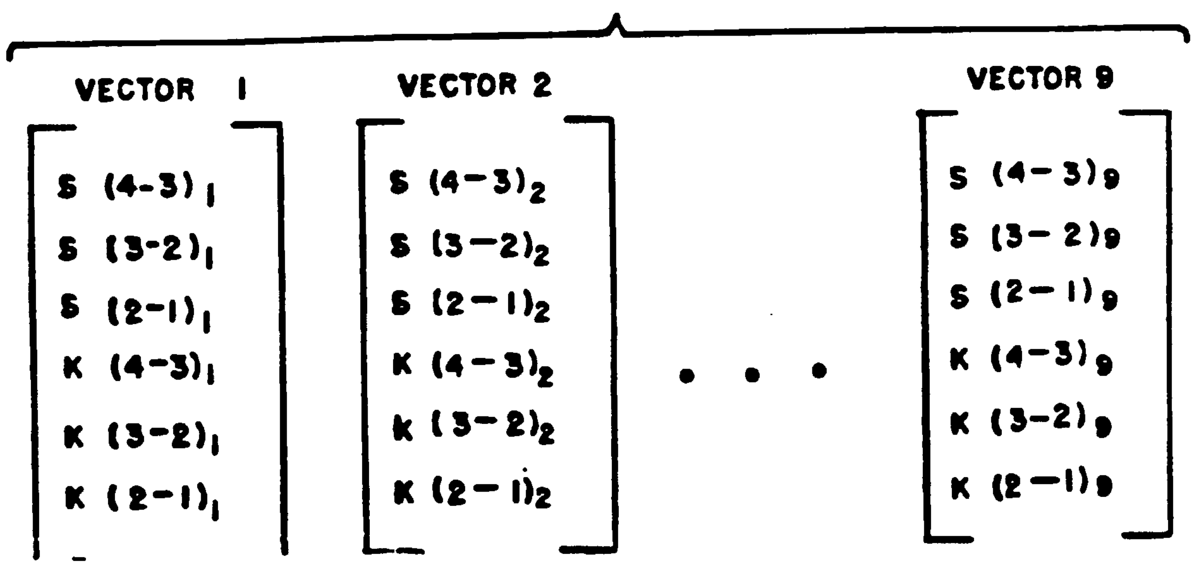

VECTOR 1
S (4-3)1
S (3-2)1
S (2-1)1
K (4-3)1
K (3-2)1
K (2-1)1
VECTOR 2
S (4-3)2
S (3-2)2
S (2-1)2
K (4-3)2
K (3-2)2
K (2-1)2
VECTOR 9
S (4-3)9
S (3-2)9
S (2-1)9
K (4-3)9
K (3-2)9
K (2-1)9

FIG. 7

LEGEND
L=NUMBER OF LEVELS
F=NUMBER OF FEATURES $N=L^F$

F \ L	2	3	4	5	6	7
2	4	9	16	25	36	49
3	8	27	64	125	216	343
4	16	81	256	625	1,296	2,401
5	32	243	1,024	3,125	7,776	16,807
6	64	729	4,096	15,625	46,656	117,649
7	128	2,187	16,384	18,125	279,936	823,543
8	256	6,561	65,536	890,625	1,679,616	5,764,801

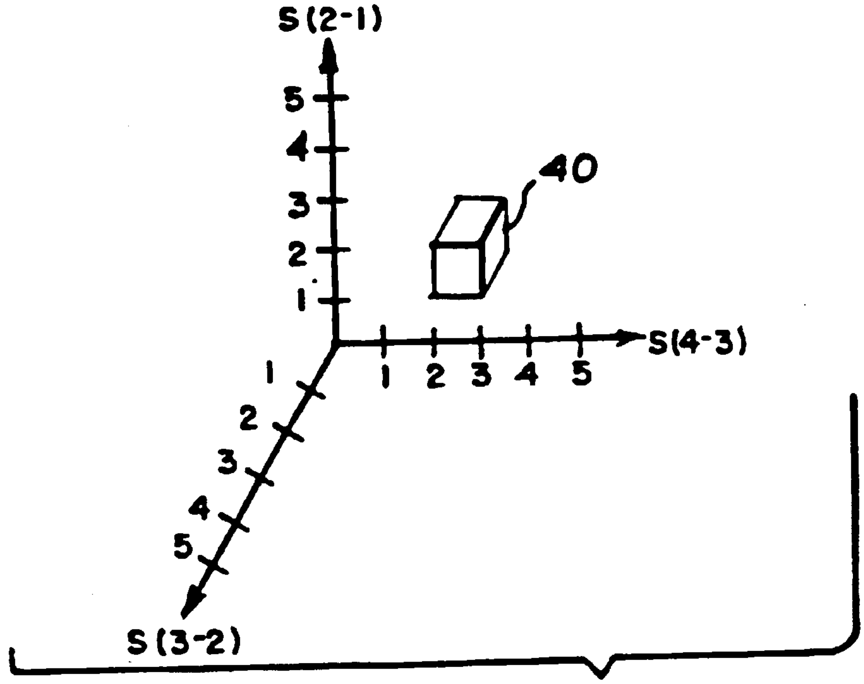


FIG. 9

FIG. 8

VECTOR 1		QUANITIZED VECTOR
S(4-3) = 0.523	→	3
S(3-2) = 0.052		1
S(2-1) = 0.229		2
K(4-3) = 0.285		1
K(3-2) = 0.981		5
K(2-1) = 0.913		4

FIG. 10

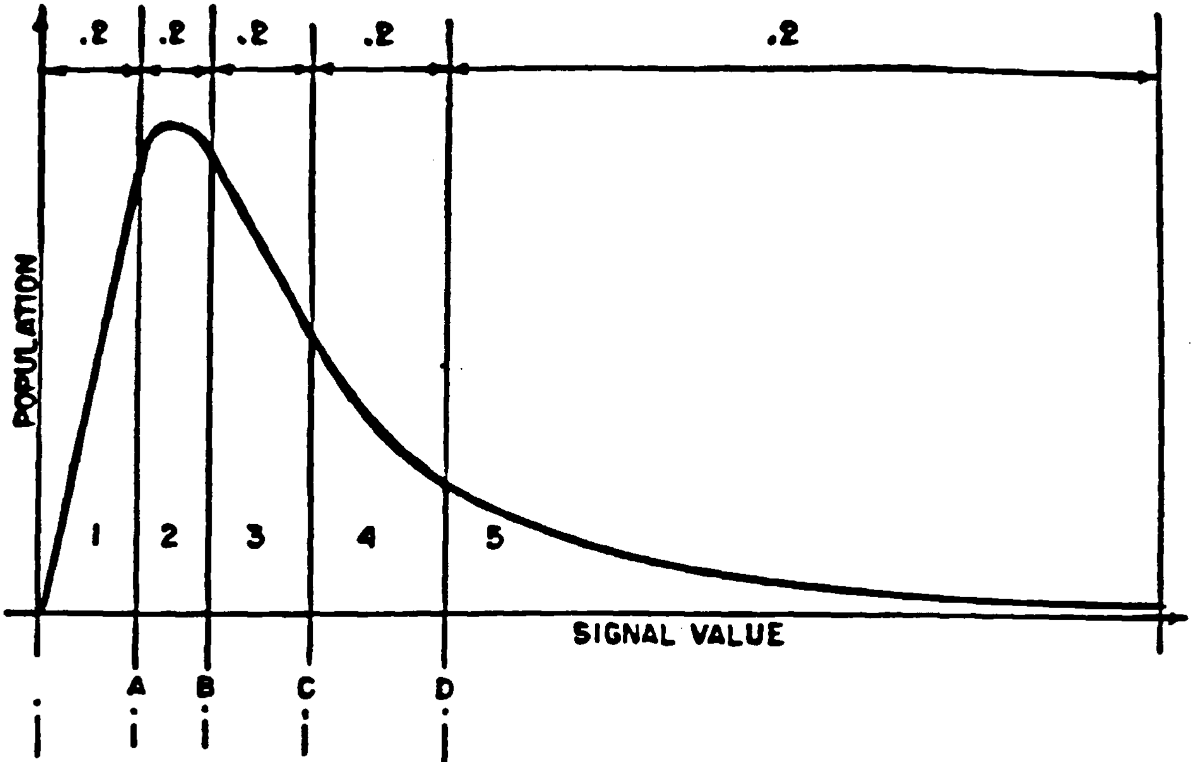
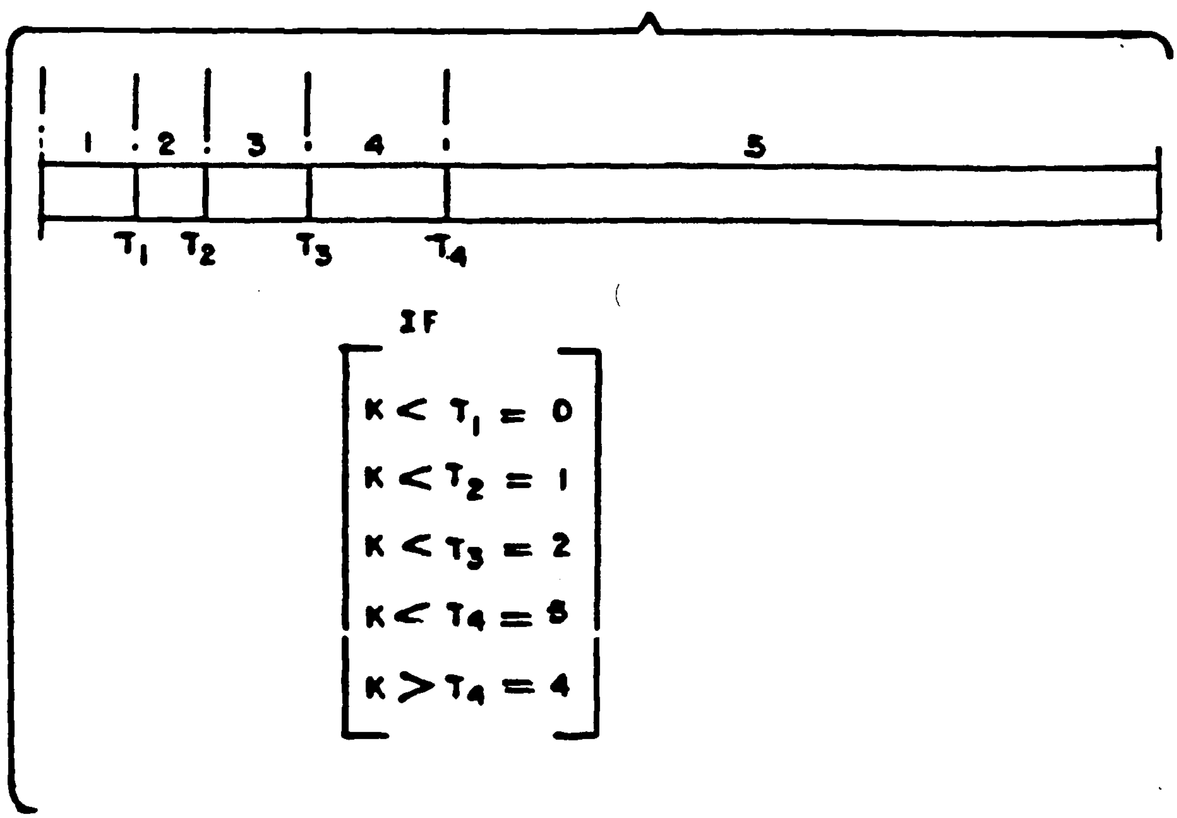
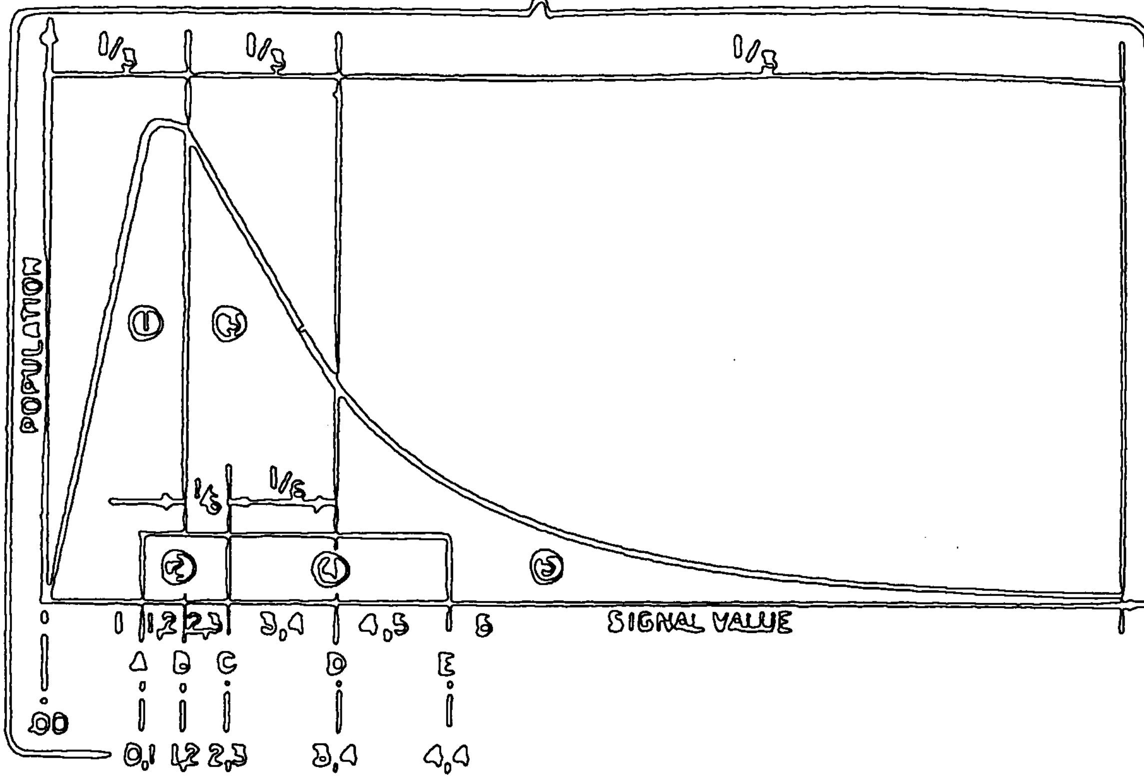
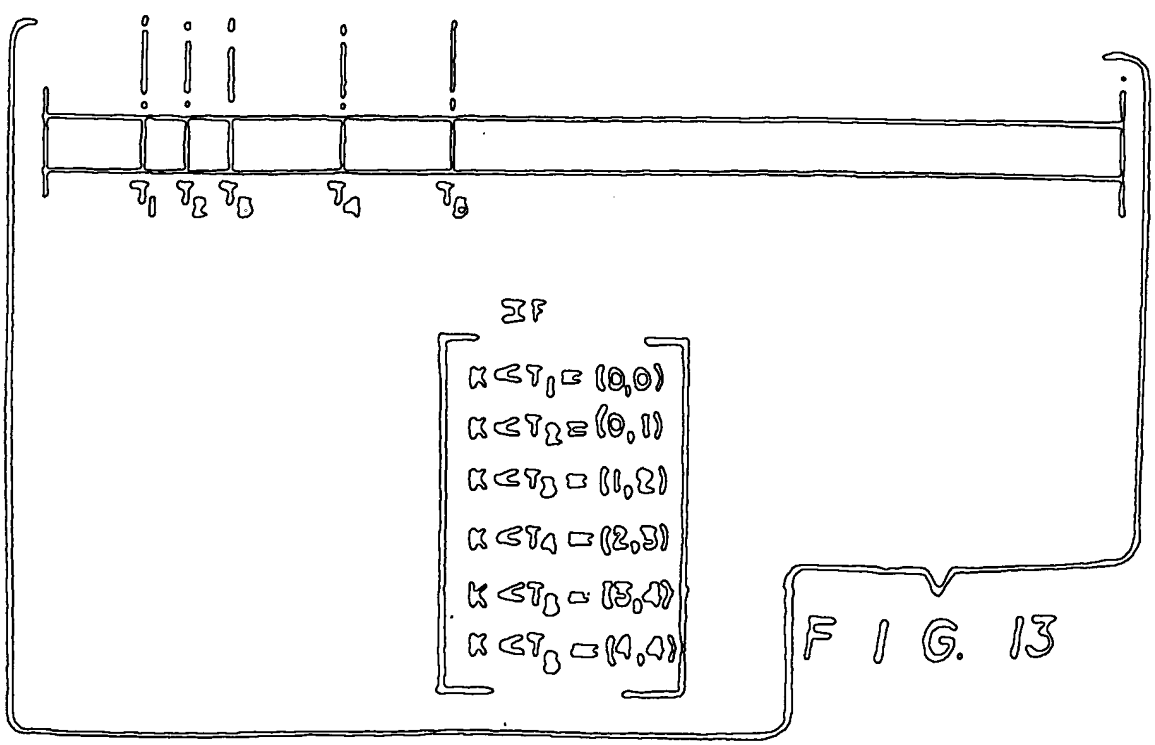

.2
.2
.2
.2
.2
POPULATION
1
2
3
4
5
SIGNAL VALUE
A
B
C
D

FIG. 11


1
2
3
4
5
T_1
T_2
T_3
T_4
IF
$K < T_1 = 0$
$K < T_2 = 1$
$K < T_3 = 2$
$K < T_4 = 8$
$K > T_4 = 4$

F I G. 12


1/3
1/3
1/3
POPULATION
1/6
1/6
SIGNAL VALUE
0
1
1,2
2,3
3,4
4,5
5
A
B
C
D
E
0,0
0,1
1,2
2,3
3,4
4,4


T1
T2
T3
T4
T5
IF
K<T1=(0,0)
K<T2=(0,1)
K<T3=(1,2)
K<T4=(2,3)
K<T5=(3,4)
K<T5=(4,4)

F I G. 13

FIG. 14

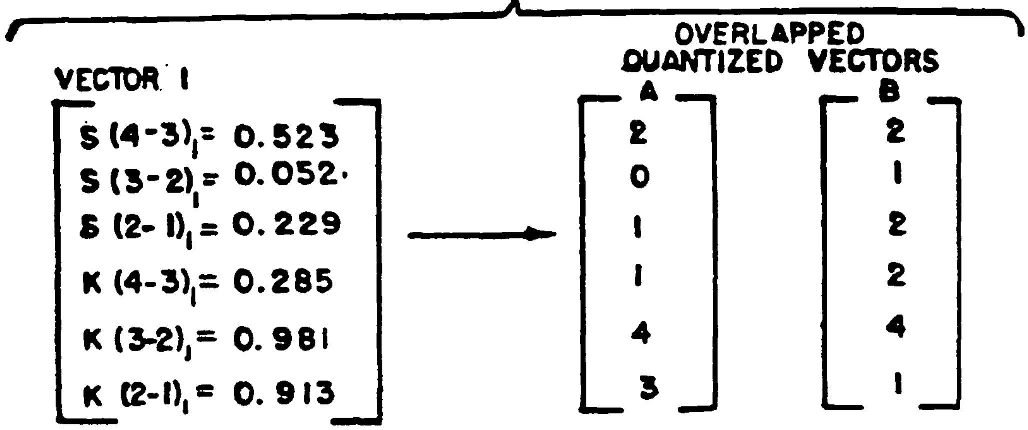
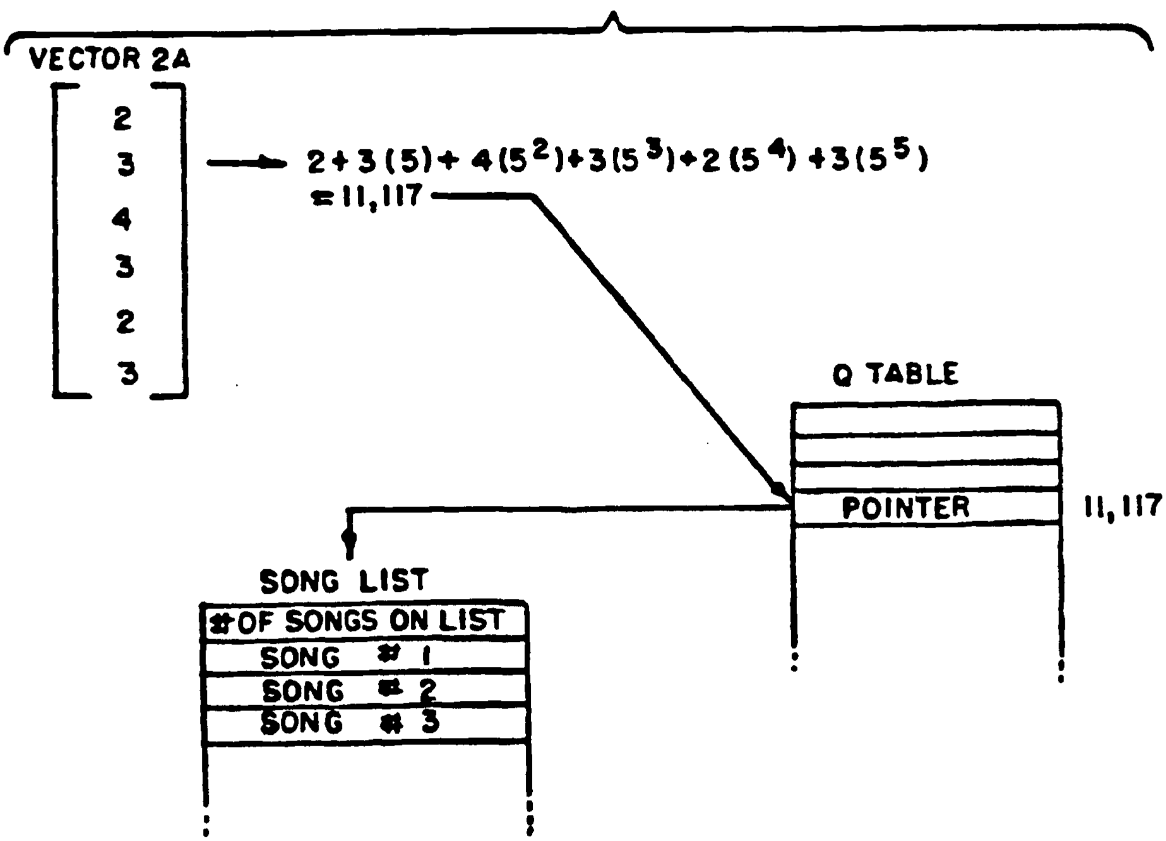
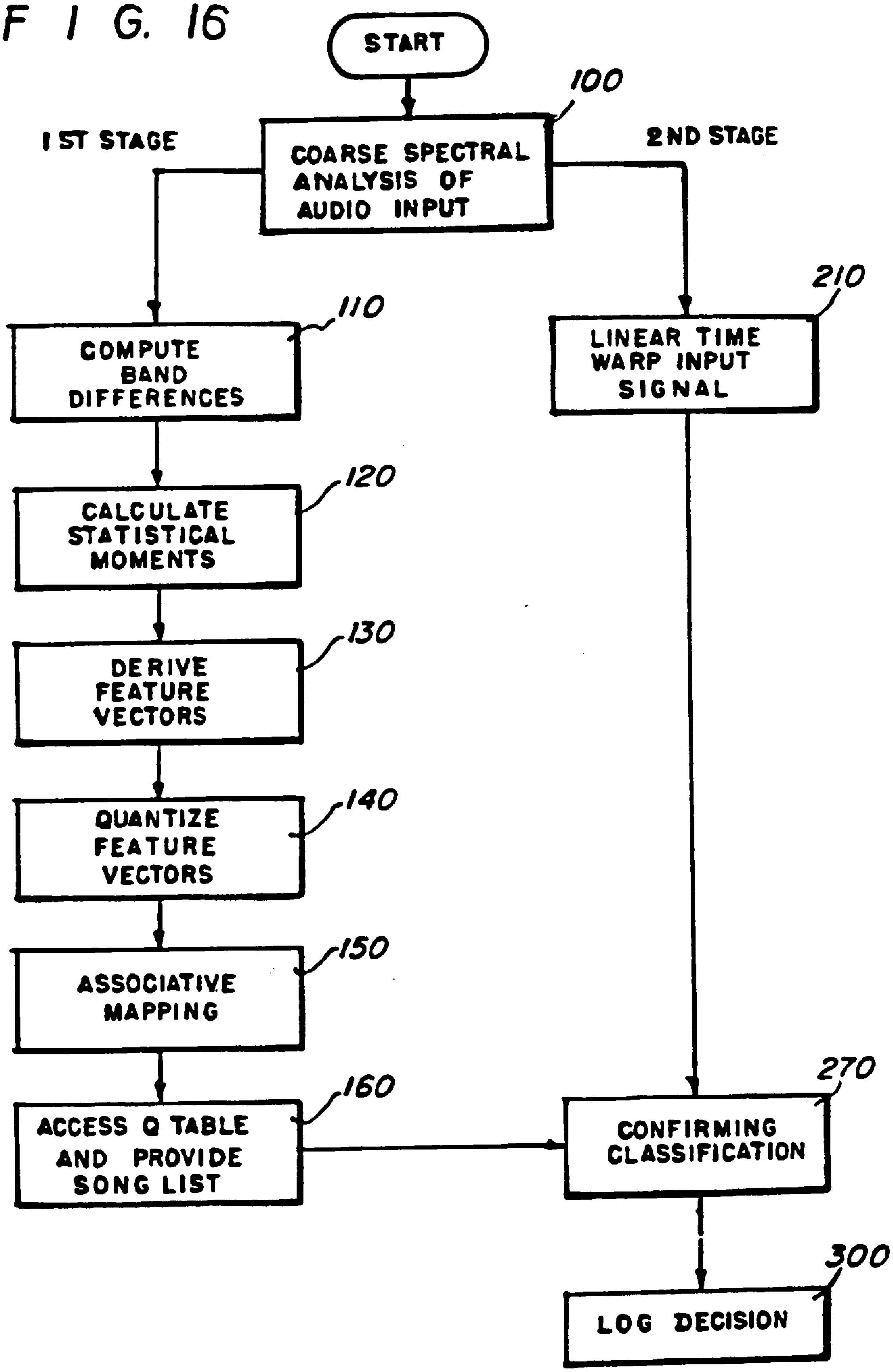

VECTOR 1
S (4-3)1 = 0.523
S (3-2)1 = 0.052
S (2-1)1 = 0.229
K (4-3)1 = 0.285
K (3-2)1 = 0.981
K (2-1)1 = 0.913
OVERLAPPED QUANTIZED VECTORS
A
2
0
1
1
4
3
B
2
1
2
2
4
1

FIG. 15


VECTOR 2A
2
3
4
3
2
3
2+3(5)+4(5²)+3(5³)+2(5⁴)+3(5⁵)
=11,117
Q TABLE
POINTER
11,117
SONG LIST
#OF SONGS ON LIST
SONG #1
SONG #2
SONG #3

F I G. 16


START
100
1 ST STAGE
COARSE SPECTRAL ANALYSIS OF AUDIO INPUT
2ND STAGE
110
COMPUTE BAND DIFFERENCES
120
CALCULATE STATISTICAL MOMENTS
130
DERIVE FEATURE VECTORS
140
QUANTIZE FEATURE VECTORS
150
ASSOCIATIVE MAPPING
160
ACCESS Q TABLE AND PROVIDE SONG LIST
210
LINEAR TIME WARP INPUT SIGNAL
270
CONFIRMING CLASSIFICATION
300
LOG DECISION

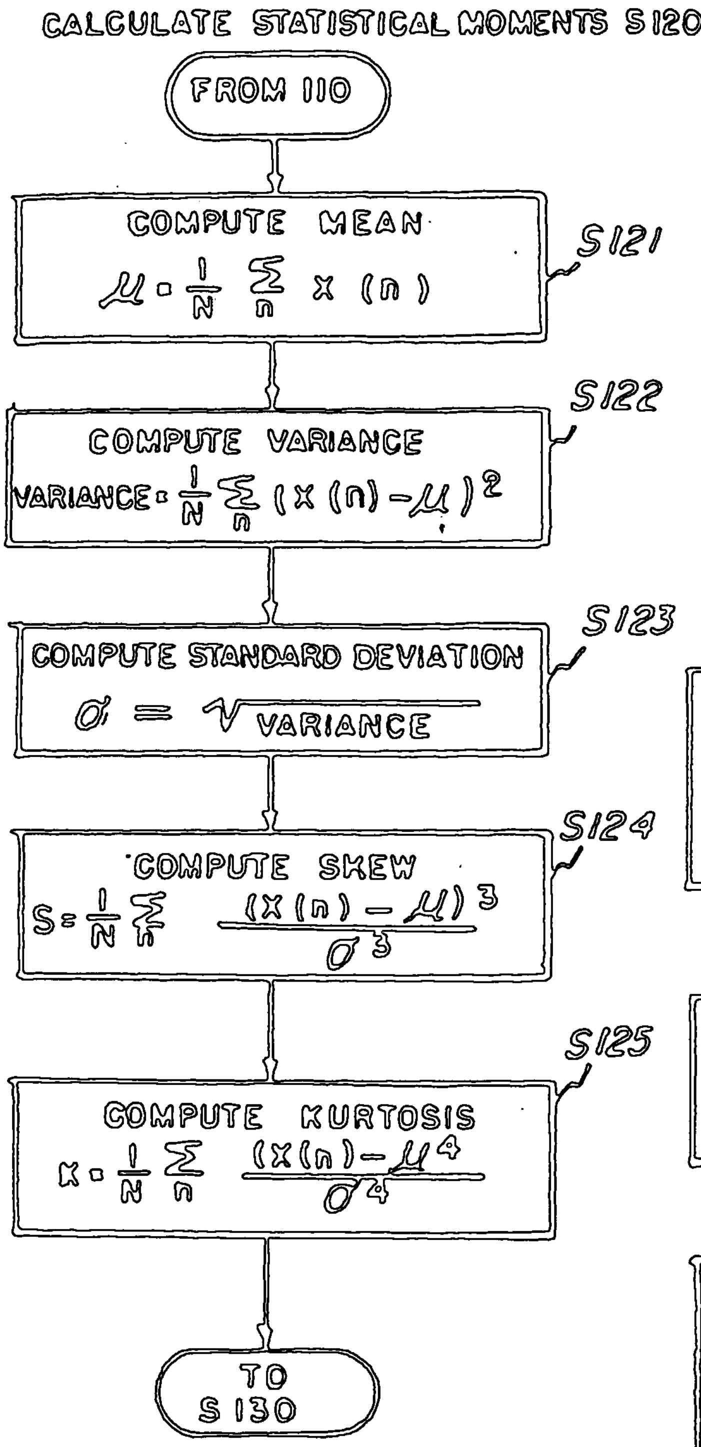
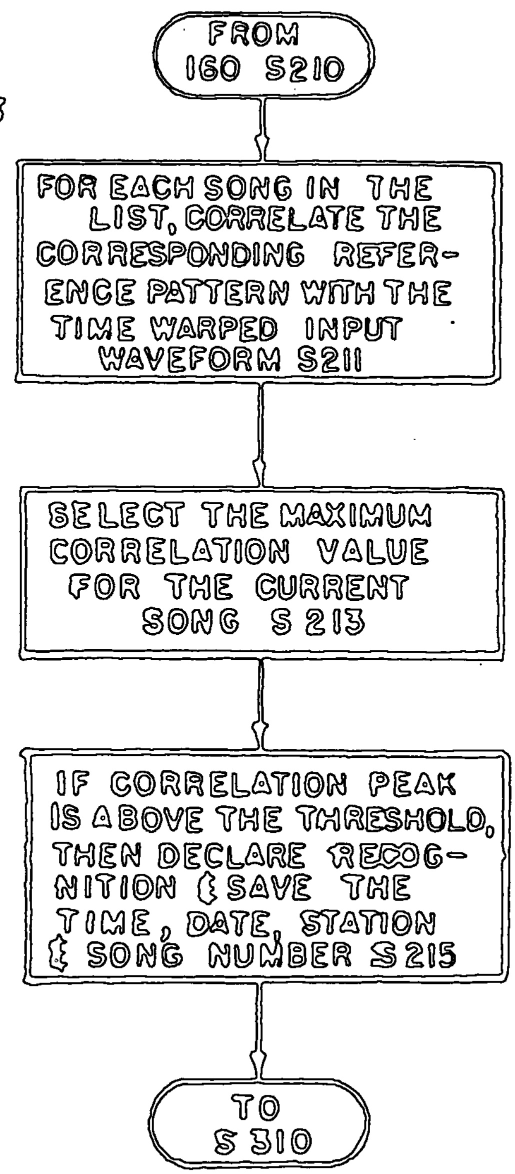

CALCULATE STATISTICAL MOMENTS S120
FROM 110
COMPUTE MEAN
$\mu = \frac{1}{N} \sum_{n} x(n)$
S121
S122
COMPUTE VARIANCE
$VARIANCE = \frac{1}{N} \sum_{n} (x(n) - \mu)^2$
S123
COMPUTE STANDARD DEVIATION
$\sigma = \sqrt{VARIANCE}$
S124
COMPUTE SKEW
$S = \frac{1}{N} \sum_{n} \frac{(x(n) - \mu)^3}{\sigma^3}$
S125
COMPUTE KURTOSIS
$K = \frac{1}{N} \sum_{n} \frac{(x(n) - \mu^4}{\sigma^4}$
TO S 130

FIG. 17

FIG. 18


FROM 160 S210
FOR EACH SONG IN THE LIST, CORRELATE THE CORRESPONDING REFERENCE PATTERN WITH THE TIME WARPED INPUT WAVEFORM S211
SELECT THE MAXIMUM CORRELATION VALUE FOR THE CURRENT SONG S 213
IF CORRELATION PEAK IS ABOVE THE THRESHOLD, THEN DECLARE RECOGNITION & SAVE THE TIME, DATE, STATION & SONG NUMBER S215
TO S 310